FEBRUARY 1985
$2.50

db

THE SOUND ENGINEERING MAGAZINE

16-TRACK REALITY

Don't just dream about owning your own 16-track. Now you can get all that extra flexibility for a whole lot less than you might think.

With models starting as low as $5900*, each Fostex B-16 ½" 16-track recorder is complete with:

- Dolby + C noise reduction circuits (defeatable) on individual record/reproduce cards
- rock stable 3-motor transport mounted on a ¼" machined alloy plate
- 15 ips with ± 15% variable speed operation
- multi-pin connector for video interlock synchronizers
- Killer Sound

Why even consider a re-built old 2" machine? At two, three, even four times the price, it won't sound as good as the B-16. And it won't even perform as well as the B-16, configured with some of the options.

For example, the model with independent tape monitoring is really a whole package:

- direct drive capstan motor with phase locked loop speed control
- 7" rack mount unit with 16 independent channels of decode & reproduce (defeat switch)
- remote control unit with individual track select buttons, headphone jack and level control, line out jack and a VU meter for fast alignment

You'd have to pay almost ten times the price of a B-16 to get this kind of dedicated monitoring function. Tape reproduce is entirely separate from the record/sync electronics.

Which makes the compact B-16 perfect for live audio and video remotes. It even has handles.

And it's as easy to use as it is to own. You can expect nice user-friendly touches like:

- blinking track numbers for record ready status
- real time tape counter with search-to-zero from either direction
- servo control over reel rocking in edit mode
- spot erase capability
- coarse and fine pitch controls with blinking LED for ON status
- optional full function remote control and auto locator

Increase your audio production capability while decreasing your costs. You'll not only save on your initial investment, but operating costs as well — both tape and maintenance.

Right now, the B-16 is the smart move in 16-track hardware. Let your Fostex Professional Multitrack Dealer° prove it. For real.

** Suggested retail*
+ Dolby is a registered trademark of Dolby Labs, Inc.
° Just call us for the nearest one

FOSTEX®
PROFESSIONAL MULTITRACK

FOSTEX CORPORATION OF AMERICA 15431 Blackburn Avenue, Norwalk, CA 90650 (213) 921-1112

Circle 44 on Reader Service Card

JANUARY/FEBRUARY 1985 VOLUME 19, NO. 1

FEATURES

COLUMNS

DEPARTMENTS

db the Sound Engineering Magazine (ISSN 0011-7145) is published bi-monthly by Sagamore Publishing Company, Inc 1120 Old Country Road, Plainview, L.I., N.Y. 11803 Telephone (516) 433 6530 db is published for those individuals and firms in professional audio-recording, broadcast, audio-visual sound reinforcement, consultants video recording film sound, etc Application should be made on the subscription form in the rear of each issue Subscriptions are $15 00 per year ($28 00 per year outside U S Possessions, $16 00 per year Canada) in U S funds Single copies are $1 95 each Editorial, Publishing and Sales Offices 1120 Old Country Road, Plainview, New York 11803 Controlled circulation postage paid at Plainview, NY 11803 and an additional mailing office

Letters

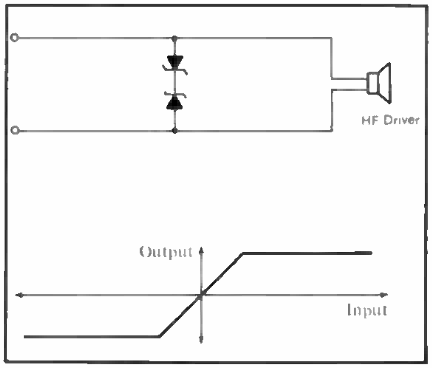
DIAGRAM ERRORS

To The Editor:
The September 1984 issue of db specializing in articles on speakers was informative and made interesting reading. However, there did appear to be a couple of errors in *Figure 3* of the article on High-frequency Driver Protection by John Eargle.

The left-hand diagram of *Figure 3* shows two Zener diodes connected in parallel. Since each diode will conduct in the forward direction on the appropriate half cycle with a voltage drop of about half a volt, it will prevent the voltage from rising to the Zener voltage of the other diode. Thus the circuit will behave as if normal gemanium diodes were used, and the voltage will be limited to about one volt peak-to-peak. The Zener diodes should be connected in series opposing. The peak voltage will then be the Zener voltage of one diode plus the forward drop of the other.

The second error has to do with the right-hand diagram. This graph shows the output voltage dropping to zero when the input voltage exceeds a certain level. This may be approximately true of certain avalanche diodes but a Zener diode voltage will stay substantially constant while the diode is conducting in the reverse direction.

Bruce A. Francis



It has been pointed out to the author that Figure 3 *in the September column of Sound Reinforcement was in error. A corrected figure is given here. Note that the Zener diodes should be placed in series opposing across the driver. Also, when the Zeners conduct, the voltage across them will stay constant rather than going to zero. We hope that this error has caused no inconvenience for readers of* db.

Index of Advertisers

About The Cover

• Sound at the XXIII Olympiad in Los Angeles. Seen here is an engineer in the audio control room for the synchronized swimming competitions. The mixers shown are the RAMSA WR-130P portable consoles. Also shown is assorted outboard gear from dbx, RAMSA, MXR, Technics, Panasonic, and Clear-Com.

You and your customers have wrestled with bulky microphones, sagging stands, and awkward cables long enough. Introducing Audio-Technica UniPoint™ cardioid Fixed-Charge condenser microphones. Perfect for the pulpit, podium, and a host of other applications. The slimmest cardioid microphones ever! Easy to mount, adjust, and use.

There are five basic UniPoint models to cover almost every application. For the podium, our double-gooseneck AT837 adjusts to any height or angle with just a touch. The AT857QM has similar dimensions but wider range and plugs directly into any surface-mount XLRF-type connector. No sag. No slip. No stand noise.

To top off a standard desk or floor stand, our AT855 is ideal. A single fixed bend puts the microphone out where it's needed. The cable is hidden inside the tube, exiting just above the stand coupler for a neat, professional appearance.

With its adjustable wire guide to set the angle, the AT853 makes a great overhead microphone for choir or orchestra without showing itself. Light in weight, cardioid in pattern, and truly tiny, it suspends on its own cable, with the electronics module high up and out of sight. Or use the included adapter on a floor or desk stand.

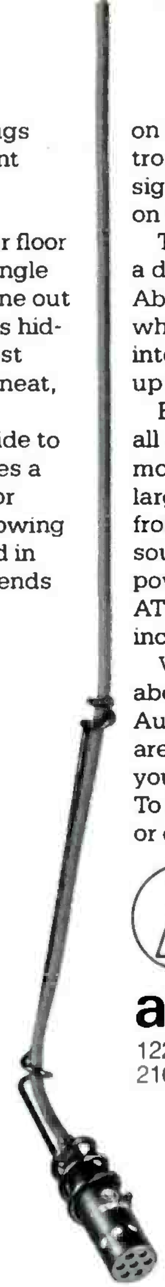
The AT859 is a wand mike with a difference: the *cardioid* pattern. About a foot long, it extends to 18" when you need extra reach. For interviews, talk shows, or to sneak up close in a news conference.

Because of their small diameter, all UniPoint microphones exhibit more uniform off-axis rejection than larger mikes. All can be powered from any 9-52VDC phantom power source, with a battery or an external power source as options. The AT853, AT855 and AT857QM also include a switchable low-cut filter.

Versatile, superb sound, and – above all – inconspicuous! The new Audio-Technica UniPoint cardioids are elegant solutions to some of your most common sound problems. To find out more, write for literature or call today.

The End of the "Klutzy" Cardioid!

audio-technica.
1221 Commerce Dr., Stow, OH 44224
216/686-2600



Circle 37 on Reader Service Card

Digital Audio

BARRY BLESSER

More Words

• This month we continue with our glossary of audio words. We are using a glossary format because many of these terms have already been discussed in detail but you may not remember their exact meanings. Last month we suggested that you give a colleague a test. How did his answers compare to the glossary of definitions?

1. CONVERSION

A process by which information is converted from one format to another. We have used this term in the context of an analog-to-digital converter or a digital-to-analog converter, but conversion could be from binary to binary coded decimal, even to odd parity, etc. The term could even be used to describe a tape recorder as a voltage to magnetic field converter.

2. A/D

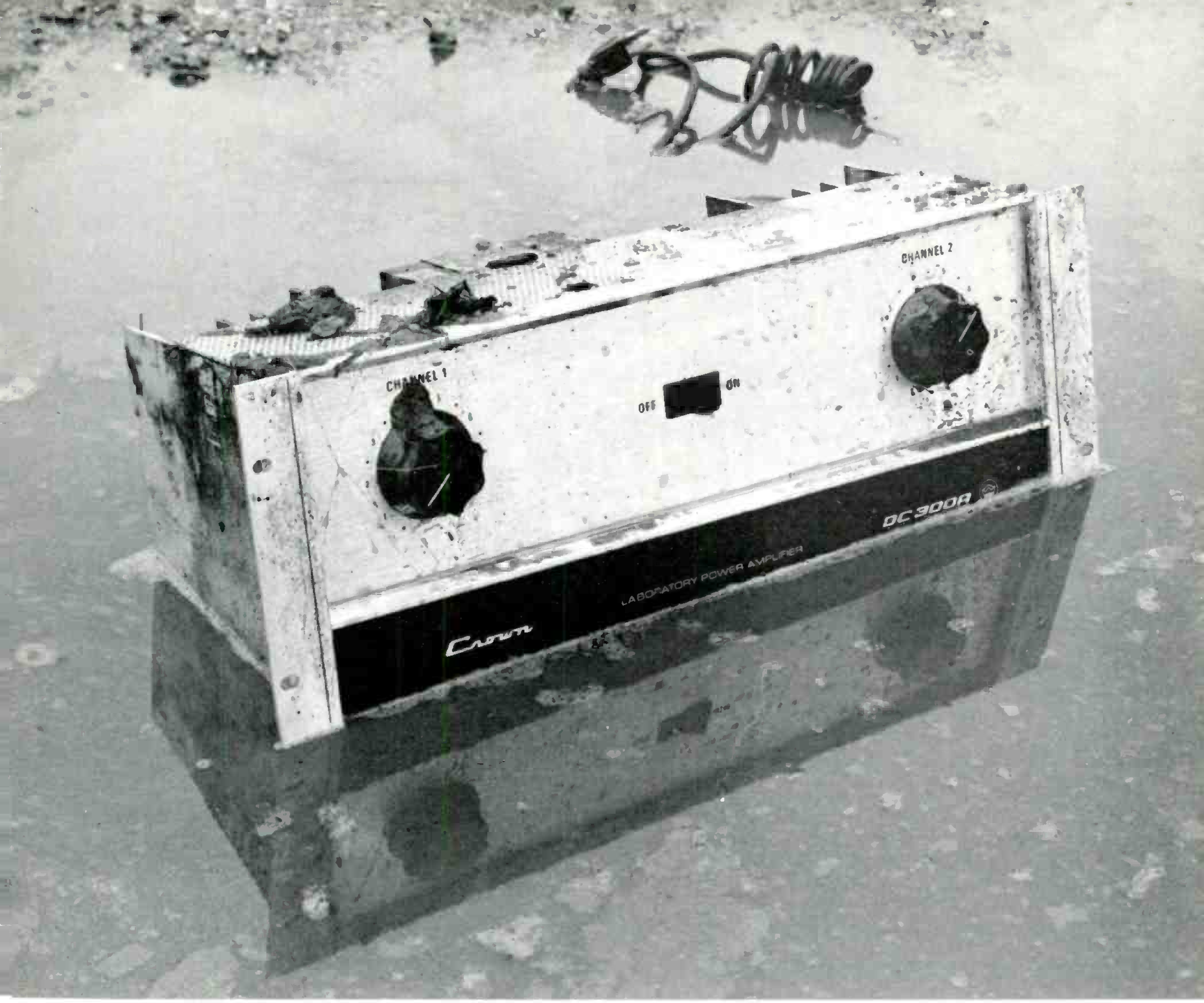
The abbreviation for analog-to-digital converter. Sometimes this term refers only to the subsection which converts a single constant voltage to a specific digital word; and sometimes it refers to the entire process of converting a continuous audio signal to a stream of digital words. In this case, the hardware would include the actual A/D as well as the sample-hold and low-pass filter.

3. D/A

This is the digital-to-analog converter and like the A/D it sometimes refers just to the subsection and sometimes to the full output structure. Specifically, the analog-to-digital converter hardware may actually contain a module which performs a digital-to-analog conversion; in system terms, D/A is the output process.

4. SAMPLING

This is the process by which a single value of a continuously chang-



In the early evening of Sept. 17, 1973, Jay Barth was at the wheel of a 22 ft. utility truck that was loaded with sound equipment. Just south of Benton Harbor, MI an oncoming car crossed the center-line; fortunately Jay steered clear of the impending head-on collision. Unfortunately, a soft shoulder caused the truck to roll two and one half times. Exit several Crown DC-300A's through the metal roof of the truck's cargo area.

The airborne 300A's finally came to rest — scattered about in a muddy field, where they remained partially submerged for four and a half hours.

Jay miraculously escaped injury; the amplifiers apparently had not.

Unbelievably, after a short time under a blow-dryer all the amps worked perfectly and are still going strong.

The rest — and the truck, is history.

crown®

1718 W. Mishawaka Rd.
Elkhart, IN 46517
(219) 294-8000

Circle 38 on Reader Service Card

ing signal is extracted and held for a short duration. It has nothing to do with digital audio since it produces a voltage continuous output. All possible voltage samples (amplitude) are still possible. Historically, this term was used for any sampled data system. Weather reports are sampled data since the weather is reported at discrete time samples; the weather itself is a time continuous process.

The act of sampling can be implemented by a sample-hold device or by some A/D converter systems directly. The very highest frequency oscilloscopes use a sampling head to effectively down-convert the signal. A digital audio system requires a sampler process but sampling does not imply digital. People eat food, but food is unrelated to people.

5. SAMPLE-HOLD AMPLIFIER

This is the name of a particular circuit technology which can implement the sampling function and hold the result constant for the A/D conversion. The hold process is dictated by the particular characteristics of the A/D process; it is not inherent. In reality, most sample-hold amplifiers are actually integrate-and-hold since they do not extract a single value from the audio signal. They extract an average over a short interval of time.

6. DE-GLITCH SAMPLE-HOLD

This is the name for the sample-hold circuit which is used in the D/A process. It has a different name because it serves a different function even if the hardware is the same as that used in the A/D. The D/A output produces a special kind of noise transient error when changing states and the de-glitch [removal of this glitch transient] disconnects the D/A from the output during this interval. Instead it holds the previous good data until the D/A comes to new, good data after the glitch has passed. It, too, may be implemented as an integrate-and-hold amplifier. Unlike the input process, the output sample-hold is not actually serving the function of sampling because the digital data [and analog counterpart] are already discrete samples.

7. SAMPLING RATE

The hardware at which the input is sampled. This is defined by the hardware. The hardware goes at its rate based on the designer's engineer choices.

8. NYQUIST RATE

This is the rate that the sampling should have in order to correctly sample a given frequency. We would say that the Nyquist rate for audio having a bandwidth of 20 to 20 kHz is 40 kHz. It is defined by a factor of two above the highest audio frequency. The actual sampling rate may be above the Nyquist rate, resulting in oversampling [more samples than the absolute minimum required]. Or the actual rate may be less than the Nyquist rate, resulting in undersampling [too few samples to sample the highest frequency]. The term Nyquist rate is a reference rate.

9. NYQUIST FREQUENCY

This is the highest audio signal frequency which can be allowed in a system with a given sampling rate. It is the inverse of Nyquist rate in that it is a reference for the audio signal. The Nyquist frequency is half of the sampling rate. The Nyquist frequency in a 50 kHz sampling system is 25 kHz even if the audio is bandlimited to 20 kHz.

10. ALIASING

Aliasing or alias distortion is the process by which an incoming frequency is converted to a new frequency. Because a sampled data system cannot represent any frequency above the Nyquist frequency, all frequencies above it are converted to other frequencies below. It is a distortion process but one which is not harmonics of the incoming signal. There is no counterpart in an analog system. Aliasing distortion is unique to sampled data processes and has nothing to do with digital.

It is usually assumed that aliasing can only appear at the input but defects in the output structure can produce aliasing. Aliasing can even be made entirely within the sampled domain. Consider a digital signal processing algorithm which contained a squaring function. This produces a spectrum which has a second harmonic. If a 22 kHz sinewave is squared, we will get a 6 kHz component in a system running at 50 kHz. The 44 kHz 2nd harmonic was aliased to 6 kHz.

11. ANTI-ALIASING LOWPASS FILTER

A lowpass filter placed at the input is designed to remove all spectrual components above the Nyquist frequency. This prevents aliasing. However, any distortion within the filter or after the filter can produce harmonics which are above the Nyquist frequency. These will alias.

12. IMAGE FREQUENCY

The output samples contain the main signal as well as spectral images around multiples of the sampling frequency. These images are inherent in the sampling process. They are unrelated to aliasing. To illustrate, in a 50 kHz sampling system, we note that 30 kHz will alias to 20 kHz and 20 kHz input will be correctly represented as 20 kHz. At the output the 20 kHz has images at 30 kHz, 70 kHz, 80 kHz, 120 kHz, etc.

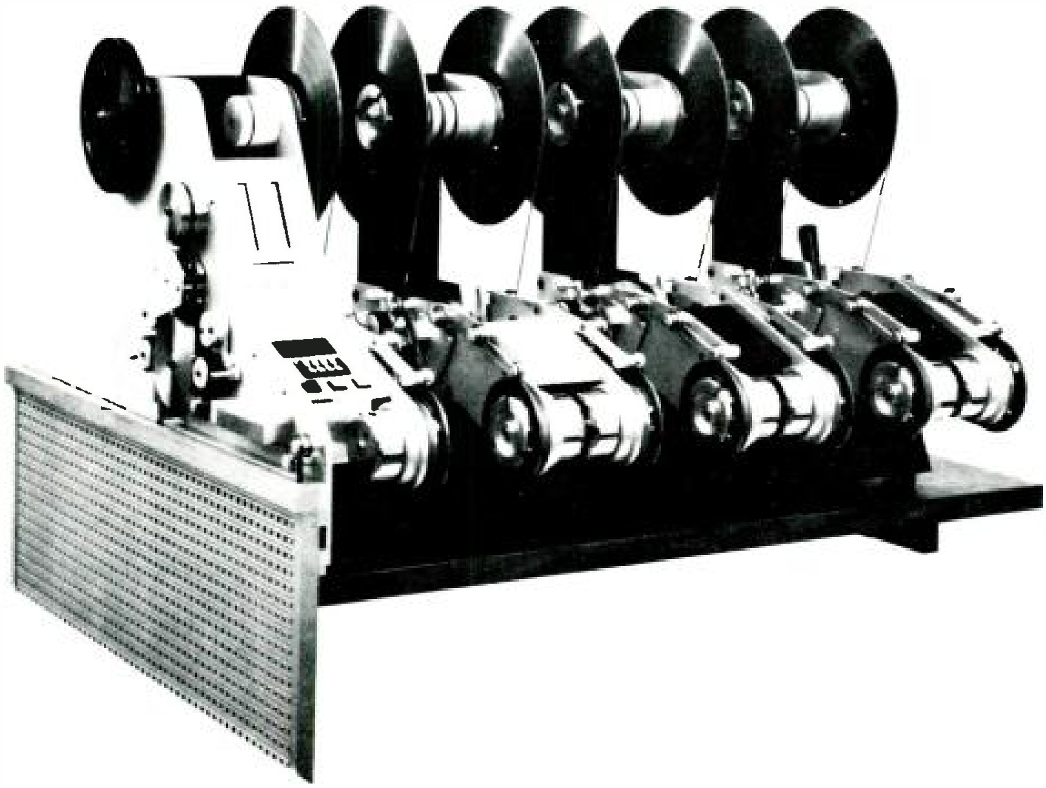
13. ANTI-IMAGE LOWPASS FILTER

This is a lowpass filter placed at the output to remove the images and to pass the main signal. Although it may be the same filter circuit as used in the input, it serves a different function and it has a different name. It removes image frequencies independently of aliasing.

We Listened To You...

For years, through our customers and market research, we have been listening in order to find out what you like and what you dislike about your tape duplicating equipment (ours or theirs) and to know what features you would include in the tape duplicator of your dreams.

The result is the 7000 Series by Magnefax



but don't take our word for it.

Listen To Us

ame________________________ Company________________________

ddress ________________________ State________ Zip________

none (_____)________________________

magnefax

International, Inc.
Route 1, Rogers, AR 72756
(501) 925-1818 TELEX 53-6433 AIDC LRK

To discover the new Magnefax, send for the whole story and a demo cassette.

Circle 39 on Reader Service Card

mediate conversion to a continuous format and then re-sampling at the newer rate. Modern technology allows for the process to be done entirely as signal processing without the intermediate step of conversion to continuous format.

22. DE-SAMPLING

This is a particular subclass of sampling rate conversion. Generally the term is referred to a process which simply skips samples. If we have a 100 kHz sampling system we can take every other sample to create a 50 kHz rate. This kind of de-sampling only makes sense when the initial samples were already band-limited to 25 kHz or less. A digital lowpass filter with a cut-off at 25 kHz could be used to insure that the de-sampling did not produce any aliasing.

23. RE-SAMPLING

This is the reverse of de-sampling. It is generally used to indicate a process where the sampling rate is increased by an integer amount. Doubling the sampling rate requires the generation of a new sample in between each of the old samples. This can be implemented with an interpolation filter (lowpass) to create the new samples. De- and re-sampling are often carried out entirely within the digital domain.

24. DITHER

To reduce the effect of granulation noise, an additional noise signal can be added to the input. Sometimes a corresponding digital version of the dither signal is subtracted after A/D conversion. Although the total noise is generally increased, the perception of the noise is improved. In specialized designs, the dither noise can be placed in the spectral guard region between the highest audio frequency and the Nyquist frequency. The term dither comes from radar technology where the mechanical gears had to be jiggled to avoid back-lash.

For the highest quality audio systems, the noise from the anti-alias filter itself can provide sufficient dither noise to prevent granulation noise. Also, the sample-hold and A/D may contain an additive noise process which serves the same function. A system with dither can allow for a sinewave which is much smaller than 1 LSB. The sinewave sounds like a sinewave rather than a square wave. This is one way to prevent a digital audio system from sounding any different than an analog system. Avoiding granulation noise is a significant issue which should never be ignored.

25. CLOCK JITTER NOISE

This is the noise created when the sampling clock is not perfectly periodic. It mostly affects high frequency signals where small time errors convert to larger amplitude errors. Random time jitter produces random additive noise but the noise is a modulation noise. It is proportional to the product of the jitter amplitude, signal amplitude, and signal frequency. The worst case then becomes a signal at the Nyquist frequency at full amplitude. In a 50 kHz system the requirement to achieve 90 dB S/N with a full 25 kHz signal means that the clock jitter must be in the sub-nanosecond region. Specialized equipment is required to make this measurement. ■

Speech Privacy And Noise Masking Systems

• Although they are not, strictly speaking, sound reinforcement systems, speech privacy and noise masking systems often make use of the same distributed loudspeaker array that may be used for paging systems. Thus, it is appropriate to discuss these systems in this column.

We have all had the experience of trying to sleep in an extremely quiet environment interrupted from time to time by discrete noises, such as a dog barking or automobile horn. The difficulty in falling to sleep under these circumstances arises from the discrete nature of the interfering noises against the extremely low background noise level.

Now, in contrast, imagine the background noise level raised by the gentle fall of rain on the roof. The new noise level of the rain is not high enough in itself to be disturbing, but it is high enough to mask the discrete sounds of dogs barking and horns honking. Thus, we are able to enjoy a pleasant night's sleep.

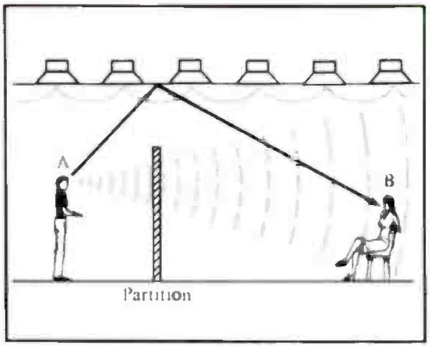


Figure 1. A noise-masking system.

This is in essence how a speech privacy and noise masking system works. Such a system in effect reduces the signal-to-noise ratio of an undesired signal to a background noise level to zero. The need for these systems comes about through the implementation of open plan office spaces in many modern commercial buildings. The basic layout is shown in *Figure 1.* A person seated at position B does not want to be disturbed by a talker at position A. In order to accomplish this, a partition is installed between the adjacent work areas, and a controlled masking noise is introduced into the area by way of a dense overhead array of loudspeakers.

Sound from A reaches the listener by way of diffraction around the partition as well as reflection from the ceiling. The sound arriving at B via these two paths is attenuated,

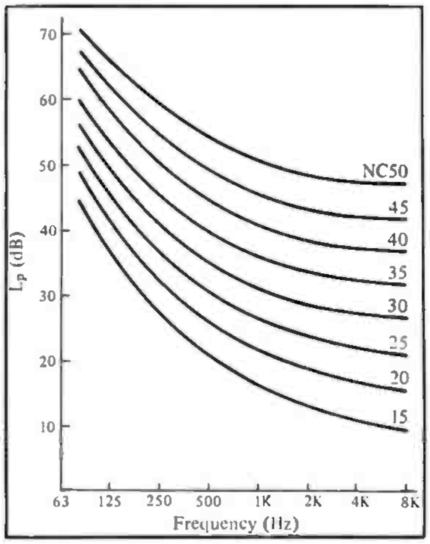


Figure 2. Family of NC curves.

especially at high frequencies, and it is low enough that it can be masked by a suitably equalized random noise signal introduced by the loudspeaker array in the ceiling. However, if the sound originating at A is too loud, the noise level that would be required to mask it would itself become a source of disturbance to the person seated at B. Conversely, if the attenuation from A to B takes place over a sufficiently large distance, and if the sound originating at A is a normal one, the partition may be eliminated.

The nature of the masking noise itself is critical, regarding both spectrum shape and level. *Figure 2* shows the family of Noise Criteria curves used for assessing noise levels in architectural spaces. They have very nearly the same shape as the familiar equal loudness contours which most readers of db have seen many times.

Most people will be unaware of a noise spectrum corresponding to NC 30. While a noise spectrum corresponding to NC 35 will be apparent to some people, most people can adjust to it. NC 40, however, is the maximum noise spectrum which can be used in practice.

STC CURVES

Having established an allowable masking noise spectrum and level which can be introduced into the space, the designer now turns his attention to the sound attenuation between adjacent work areas. We now examine the curves shown in *Figure 3.* These are known as Sound Transmission Class (STC) curves,

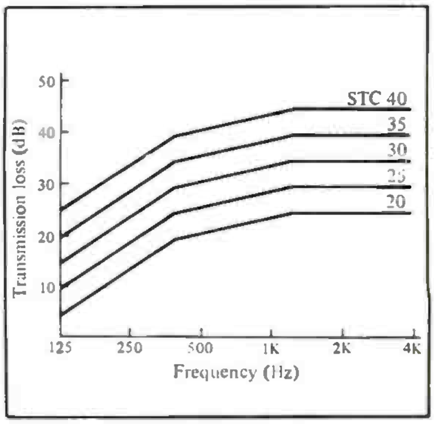


Figure 3. Family of STC curves.

although in this application we may refer to them as Noise Isolation Class (NIC) curves. Fortunately, most acoustical barriers are more effective at high frequencies than they are at low frequencies, so that their transmission loss with respect to frequency is roughly the inverse of the equal loudness contours. A barrier or transmission path rated at STC 20 will introduce a loss as indicated by the corresponding curve.

STC 20 represents the minimum

loss, referred to one meter, the should exist between A and B for successful masking, taking into account all of the acoustical paths between A and B. This consideration usually requires that the distance between A and B be no less than 3 to 4 meters (10 to 13 feet) and that partitions with a transmission loss between 400 and 2000 Hz of not less than 10 dB be used. The surfaces of the partitions should have a fairly high absorption coefficient, and they should extend to the floor. The partition height is a design variable that may be dictated by the degree of privacy required.

It is generally felt that the sum of NC rating of the masking noise and the STC rating of the transmission path be equal to about 60. In making this determination, a sound source at A is measured at a distance of one meter at one-third octave intervals over the range of 400 to 2000 Hz. Then the same measurements are made at B. The differences represent the interzone attenuation between A and B. The specific noise masking spectrum may then be shaped so that, on a one-third octave basis, the sum of the STC ratings and the NC ratings is equal to 60 or greater.

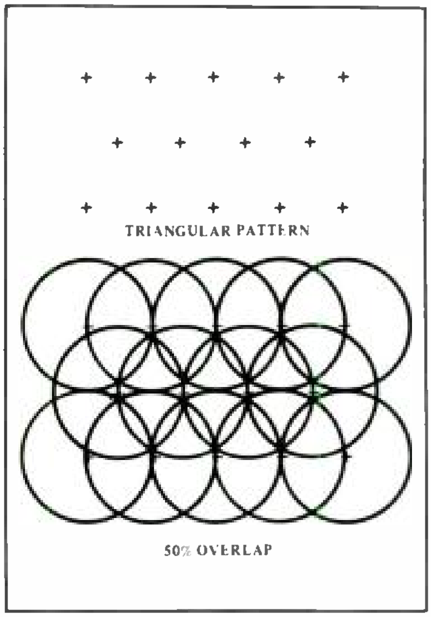


Figure 4. Loudspeaker array for densest coverage.

We hasten to state that the specification and final adjustment of systems such as these require much expertise, and these are matters best left to qualified acoustical consultants who have had considerable experience in the area. Among the important considerations which would be addressed by the consultant are:

1. Density and nature of loudspeaker array. (*Figure 4* represents the desired triangular array which produces the most even coverage. Multi-channel noise sources have been specified for greater diffusion of the masking noise. For masking-only systems, loudspeakers facing upward in the plenum above the dropped ceiling have been used.)
2. Electronics specification. (The high crest factor of random noise requires considerable electrical headroom. Individual loudspeaker levels have to be carefully adjusted, and system stability is essential.)
3. Acoustical treatment and spacing of work areas. (The precise acoustical nature of partitions and ceiling materials must be specified, and work spaces must be located appropriately.)

Obviously, the specification of one of these systems is a job for a consultant with considerable background and experience. Our discussion here has been only an overview of a complex art. ■

JESSE KLAPHOLZ

Computers: The Paradoxical Infancy

• In case you haven't noticed by now, my outlook on the general use of computers may at times seem to be a bit dismal. Don't get me wrong, I'm just as much an advocate of digital computing as I am against such practice. These views are the result of continually seeing the business world attempting to use small computers when the manufacturer of the system itself will be the first one to say, "We never intended this system to be used for this application." On the other hand, I have seen engineers get so wrapped up in the design of a system intended to make the design/manufacturing process of their products simple, that they end up with an incredibly complicated and ineffective process.

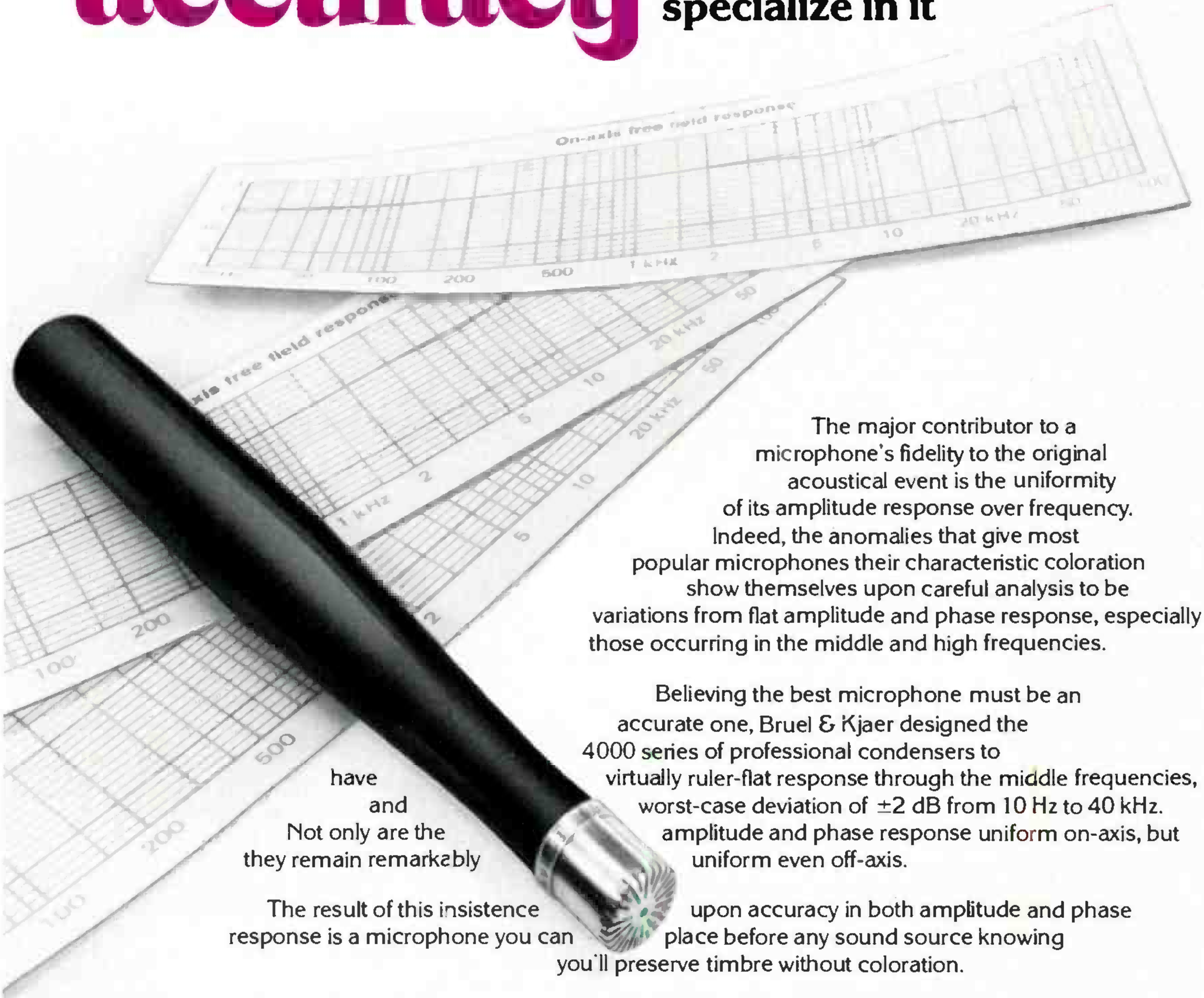
I can't put it any better than Al Charpentier, the designer at MOS Technology of the 6502 IC chip (the CPU of the Apple and Commodore computers). "Our job is to design and develop products, and not to divert our energies in the design of tools to do our job." It is somewhat disappointing to see that so many of us are becoming so dependent upon machines that we are thinking less and less. For example, how many times have you waited for a store clerk to make change because they can't even subtract 98 cents from a dollar without the machine telling them what to do? However, I will have to admit to occasionally catching myself using my religiously worn wristwatch/calculator for mathematical computations that one had to do mentally, once upon a time, in order to graduate high school.

Man can think out problems, make observations, and come up with creative solutions in such 'high-level' approaches, which are still a far cry from what we presently are capable of implementing on our personal computers. The power of the human mind is often diluted by the 'crutch' that the use of the personal-computer in our everyday lives has become. It is mind boggling that DaVinci's "flying machine," designed to give 'wings' and enable a human being to fly under his own power, was not perfected until some time in the third generation of digital-computers—after we had already landed on the moon!

Similarly, the chief design engineer of the Apple Macintosh, practically designed the entire Macintosh circuit board in his head going through several proto-type development stages. For those who aren't familiar with the inner workings of the Macintosh, it has half the parts of 64K IBM PC Jr., yet it features 128K of memory, printer, modem, and serial ports, and has its disk-drive

A few words on microphone accuracy

from the people who specialize in it



The major contributor to a microphone's fidelity to the original acoustical event is the uniformity of its amplitude response over frequency. Indeed, the anomalies that give most popular microphones their characteristic coloration show themselves upon careful analysis to be variations from flat amplitude and phase response, especially those occurring in the middle and high frequencies.

Believing the best microphone must be an accurate one, Bruel & Kjaer designed the 4000 series of professional condensers to have virtually ruler-flat response through the middle frequencies, and worst-case deviation of ±2 dB from 10 Hz to 40 kHz. Not only are the amplitude and phase response uniform on-axis, but they remain remarkably uniform even off-axis.

The result of this insistence upon accuracy in both amplitude and phase response is a microphone you can place before any sound source knowing you'll preserve timbre without coloration.

If you like performance curves, request our literature. If you'd like a demonstration in your space, call your B&K field applications engineer or contact:

B K

Bruel & Kjaer Instruments, Inc.
185 Forest Street
Marlboro, MA 01752
(617) 481-7000
TWX: 710-347-1187

Bruel & Kjaer (UK) Ltd.
Cross Lances Road
Hounslow, Middlesex TW3 2AE
01-570-7774
TELEX: 934150 bk uk g

World Headquarters: Naerum, Denmark. Sales and service in 55 countries around the world.

Circle 36 on Reader Service Card

controller and video-monitor circuits all on a 1 ft. square circuit board. All this was accomplished for the most part without the aid of computers—the old fashioned way.

POSSIBLE POTENTIAL

When it comes to the computer 'understanding' what is going on in an electroacoustical world, perhaps we had better stop and think about this for a moment. We have years upon years of intensive 'programming' in our heads, starting from our early childhood years. It would be a monumental task to program into a pile of chips the psycho-physical phenomena of acoustic experiences that we are just now beginning to understand.

Those of us who are so caught up in computing for the sake of computing may not realize that personal computing is in its infancy. Computers are great tools when used as tools, but to get into personal-computing at this stage in the game, "just to keep up with technology," can be as big a mistake as it can be a great learning experience. What we think is a standard today will inevitably be obsolete tomorrow. To learn how to hand crank-start an engine or work a manual choke was OK when those conveniences were needed at the time, but these skills don't apply to automatic starters and chokes of today's cars.

There is great movement behind the scenes in the computer industry as we all know. The environment as we know it in today's computers is on the whole a machine oriented one. Scientists have long failed to recognize the importance of visual thought and the mind's tendency to 'multi-task.' Until now, computers have been used to analyze numbers and manipulate text for applications such as spreadsheet calculation, word processing, and data base management. Many people are now finally realizing the visual potential of computers. However, charts, graphs, and games have only begun to take advantage of computers' visual graphic capabilities.

COMPUTER COMPREHENSION

Nature has given us two complementary processing systems. One system perceives the parts, and the other recognizes whole patterns. The one side of our brain is particularly efficient for producing and understanding language; the other side specializes in extracting meaning from the mass of information that floods our senses. Working together, these two processing systems provide us with a remarkable set of mental tools. As hardware and software designers explore new avenues of emulating our mental processing systems, they will redefine what computers can and should be capable of accomplishing.

For some time now, there's been a lot of talk about talking computers. Hot stuff hardware and software experts now claim that their machines will soon be able to *speak* in conversational English and that computer users won't have to use keyboards, mice and computer languages to get the computer to work for them. Instead, all they'll have to do is lean back and tell the computer what to do with their voice.

At the very least, they say, the computer will respond to typewritten English-like dialog and instructions. Progams such as Microrim's "Clout" and Artificial Intelligence Corp.'s "Intellect" are examples of this kind of query-language. This kind of research gives us some insight as to where the future of the computing world is heading.

On the *earthy* side of things; remember the *Whole Earth Catalog*, which began publishing its editions in 1968? Publisher Stewart Brand now brings us, *Whole Earth Review: Tools and Ideas for the Computer Age*. "For the last couple of years computers, especially personal computers, have been touted as the bearers of salvation," writes Brand in his introduction to the *Whole Earth Review*, and we bought it. And significant salvation occurred. Now that we've been saved, it's not too early to inquire about the real price." "Computers As Poison" is the cover headline on the premiere issue, dated December 1984-January 1985. But the magazine is not a total indictment of computers. The second half of the Whole Earth Review is written for people who like computers—"the true believers," as Brand calls them.

HIGH TECH VS. HIGH TOUCH

Following along somewhat of a more conservative approach in comparison to the *Whole Earth Review* is John Naisbett's Book, *Megatrends*. *Megatrends* is a well researched objective outlook on our 'high-tech' society that talks about the 'high-tech—high-touch' world we live in today. In this rapidly increasing rate of new technologies being introduced, we are losing 'touch' with some of the real things in life. That is, the more high-tech we get, the more we seek something natural and less mechanical to identify with.

A perfect example of the 'high-tech—high-touch' coexistence is Coleco Industries, manufacturer of the Adam computer. It is widely expected by the 'computer business experts' that Coleco will discontinue the Adam after Christmas. But Coleco will find solace from the holiday hysteria over its 'high-touch' Cabbage Patch Dolls, which analysts predict will account for more than $4 of every $5 in the company's revenues this year. It is clear that 'high-touch' is responsible for keeping Coleco out of the red this year.

We are all "married" to the physical laws of science in our everyday lives, both at home and as sound-people. These physical laws are so intertwined and complex that we are still proving, disproving, and developing new theories—in an analog fashion. To live with all these electro-acoustical laws of science applied strictly in a digital/linear-world at this point seems a bit premature, but perhaps not too far off in the future. ■

LON NEUMANN

The First Digital Broadcast Via Satellite! "Stevie Wonder Comes Home"

Here author Lon Neumann discusses why the "Stevie Wonder Comes Home" presentation for Showtime and Westwood One accomplished quite a few firsts for digital audio programming.

Dave Hewitt at API console in Record Plant's Black Truck.

SHOWTIME AND WESTWOOD ONE'S collaborative presentation of the "Stevie Wonder Comes Home" show on June 18, represented the first time in history that a fully digital audio soundtrack was simulcast with video. In the process of bringing this production to completion, several other firsts were accomplished. It was the first time that a fully digital audio program was produced to accompany a commercial TV show. It was the first time that digital audio was edited digitally to conform to a video "Edit Decision List." It was the first time for digital audio "sweetening" of a TV show and the first time that a stereo digital audio program was commercially distributed by a digital audio satellite system.

Lon Neumann is a recording engineer based in Southern California.

REMOTE RECORDING

The show was produced by Ken Ehrlich and directed by Walter Miller. A series of three-hour concerts was taped at Detroit's Masonic Temple. The theme of the show was to have Stevie returning to the place of his roots to perform a special homecoming show for the people who first recognized and supported the young genius.

There were to be two very welcome setup days scheduled before the first performance Thursday, April 12. This was going to be a big show. The first setup day was reserved just for staging and lighting. It was being lighted for national TV, not just local performance. There certainly is no substitute for the acoustics of a well-designed and well-appointed room! This room was a fine example of good acoustics, complete with a proscenium arch stage, upholstered seating, stunning chandeliers, heavy velvet drapery, and thick carpets. The volume and

shape of the room were very nice, too. Not too small, not too large. The bottom line is that the room sounded great.

Recording facilities were to be provided by two trucks. Unitel provided the video recording facilities and the Record Plant New York provided their black truck with Dave Hewitt and his crew for audio recording.

Audio was recorded on two 3M 32-track DMS digital mastering systems from Digital By Dickinson. SMPTE time code was recorded synchronously on one track of each of the digital machines, as well as on each of the six video machines, so that they could later be synchronized for all post-production work. SMPTE time code was fed to us from the video truck. The code was printed longitudinally on all video machines on track 3. On tracks 1 and 2 went a production work mix.

EDITING

The show was first edited together by the director using essentially visual criteria. It was then left up to me to make all the assorted audio edits work.

I had decided to do electronic digital editing of the multi-track tapes before attempting to mix down. There were several advantages to editing before mixing. When a troublesome edit was encountered, there were still several options open. For example, the whole edit point could be slipped either earlier or later than the video edit with great precision. Or, perhaps more importantly, individual tracks, or groups of tracks, could be slipped when it helped the edit to work. In other words, certain instruments could be allowed to decay out naturally from the out-going segment, while the breath before a vocal phrase could be allowed to happen early, as it would have naturally. This could neither have been done by editing after mixing, nor with razor blade editing of multi-tracks.

Electronic editing, in and of itself, had many advantages in a production of this kind. One advantage was that the editing was done without any worry of endangering the original tapes. The way it worked was that a master reel was compiled on one machine from playback of the production reels on another machine. No physical cutting of the tape ever took place. Edit points could be worked with to my heart's content. Any given edit could be done on a trial basis. If it didn't work on the first trial, it could be done again and again until it was right. That sort of freedom would never have existed if I

Showtime's video tape operator, Pat O'Grady, at the Sony BVH-2000 1-in. video tape machines.

had physically cut the master tape. There ends up being much greater accuracy with electronic editing, too. It's very easy to try the edit just a little bit earlier, or just a little bit later. In our case, being digital, the master reels were built of bit-for-bit accurate replicas, or clones, of the production segments.

There was also the matter of matching around "pull-ups." "Pull-up" is a video term referring to cutting out segments in the midst of a tune. The original performances were all about three hours long. The final product needed to be one hour long. There were a great many pull-ups to be done in the course of editing. If it had been edited after mixing, they would have run into trouble when a pull-up was done. It would not have been known while mixing whether the mix before the edit point would match the mix after the edit. If it was discovered that the edit wouldn't work, it would have to be remixed. This would have gotten to be very costly and time consuming. And, even then, there still would have had no guarantee that it would match until it was tried again.

After the video was edited, there is a computer printout that is known in video jargon as an "edit decision list," indicating the production time code and the master time code of the "in point" and the "out point" of each cut in the show. Since the video edit had been done with Complete Post's computer, it was an easy matter to do a printout of their edit. The edit decision list was four single-spaced pages long.

In addition to the pure scope of the edit, there was the problem that the 3M editor had no means to relate to SMPTE time code. What was needed was a digital editing system that would allow editing data to be entered as SMPTE addresses. This required some development. As it turns out, Q-Lock has a very nice system for doing video post-production work. This works fine, as long as you are using analog machines. So what we did was use the best of the Q-Lock, and develop the rest that would be needed to do fully digital editing to an edit decision list. Actually, the Q-Lock would have worked just fine, even with the digital machines, if I had been content to convert the digital signal to analog, and then convert it back to digital again. In other words, it could have been edited from one machine to another using an analog signal via the converters. But I was determined that I should not have to resort to that. That would have defeated one of the basic advantages of using digital technology in the first place.

At every stage of post-production, we had picture locked to audio. I can't stress enough the importance of having the picture to watch while doing audio for a video project. The final product will just never sound right if you don't watch the picture while you are doing it.

Once we had the system up and running, I still had to work my way through those four pages of edits. What I had was far from automatic, but it was digital editing, and it was to time code. But, consider this...On any given edit, what had to happen was that a piece of program must drop in precisely at a specific master SMPTE address. This master code was continuous new code that was established at the time of the video edit. But the piece

that would be dropping in would be identified by the original production SMPTE code. There would, of course, be a substantial difference between these two codes. That difference is known as an offset. This offset first had to be calculated. Once the offset was known, it was used to adjust the addressing of the slave machine controller, so that when the synchronizer had the two machines locked up in apparent sync, the precise in-point of the program on the slave would occur exactly after the last frame of the preceding outgoing edit on the master machine. The master machine would then go into record at precisely the right spot. The master machine would then stay in record for the duration of that particular edit, while the machines stayed in sync. At the time the out point was reached, the master machine would be instructed to go—out of record at an address that had been derived by once again calculating another offset between master SMPTE time code and production SMPTE time code. Thus, when this specified address was reached, the last recorded frame would line up perfectly so that the following frame would remain undisturbed with the correct positioning. This, therefore, also allowed that insert edits could be done with wild abandon.

It was imperative that all edits play back with neither disturbance to the synchronization of audio to picture, nor disturbance to the internal digital system synchronization. It's a good practice to always pre-stripe the master reel with continuous master time code, and never to punch in on this code. This pre-striped code was then what all the audio edits referenced to.

MIXING

When it came time to mix, the benefits of having edited the multi-track tape became more fully appreciated because it was already known which edits were troublesome. Any necessary corrective measures were then taken while mixing.

It was a true delight to mix the show pre-assembled. It made for much better continuity than could have otherwise been the case.

I'd like to say a few words about the ambience in the mix. I used the Calrec Soundfield Mk III system. One very distinct advantage this system has over other microphones is its ability to steer the image, after the fact. At the mixing stage one still has complete control of the image. One disadvantage other stereo microphones have is what happens when you hang them from their cables. With the passage of time, the lay of the cable tends to stretch out, causing the mic to turn. The Calrec is totally non-critical in this regard. As the mic happened to turn a little bit after a couple of days, it was easy to just steer the image back into place, while I was mixing. One thing that is required to make this system work properly is stable phase coherence in the tape machine. The Calrec does work with analog tape machines. But digital audio provides absolutely rock-solid phase response, track to track, at all frequencies. Analog doesn't even begin to compare. A coherent stereo signal like this is another case where you can really hear the superiority of digital audio.

SWEETENING

"Sweetening" is the video term for enhancing the sound track by adding audience reaction. Sweetening seems to be unavoidable with TV shows. It is usually done in the audio department of video post-production facilities. Carefully done, tasteful sweetening can enhance a show. Unfortunately, no video house is normally equipped to do

RCA transmission antennas.

digital audio sweetening. A consultation with Kelly Kotera of Compact Video revealed that they were not only able, but also very enthusiastic to cooperate with us in doing the first-ever digital audio sweetening date.

The segments that were to be added had been taken from various spots in the show, and digitally cloned across from the 32-track to a digital 4-track. Then, at the appropriate spots, the 4-track was run wild, and the respective audience tracks were then bounced back across the mix on the 32-track. We "sweetened" only with tracks recorded at roughly the same time as the rest of the live tracks, so they were a perfect acoustic match.

Sweetening did enhance the show. Happily, we didn't have to resort to using sweetening to try to conceal any mistakes. Rather, sweetening was used as a purely creative tool. Not having to bounce through several analog generations, nor having to fly in sound effects carts, kept the whole production that much cleaner.

ASSEMBLY

The sweetened mix was then transferred to the Sony PCM-1610 for final assembly. The Sony DAE-1100 digital audio editor was designed with the ability to enter edit points as SMPTE addresses. It was also possible to trick the editor itself into synchronizing the digital audio to picture.

LAYBACK

Once the entire show was assembled on to one ¾" tape, we were ready for video layback. Even though the

Stevie Wonder in concert at Detroit's Masonic Temple.

premier broadcast of this show would be with digital audio simulcast, Showtime specified that their product must be delivered with stereo/audio on tracks 1 & 2 of the 1-in. C-format video, with mono audio on track 3. Timecode was inserted in the vertical interval.

The stereo audio was laid back onto tracks 1 & 2 with Dolby A. No limiting was necessary. The combination of Dolby and judicious level setting handled it quite nicely.

The important point to note here is this, simply that digital audio was laid back to conventional video. This fact has many important implications for the *music video market*.

BROADCAST

The simul-cast was an engaging project in itself. Nothing like this had ever been done before. Showtime, Westwood One, and Sony were all as interested as I was that this event be a success.

We met a week ahead of time at the Sony plant in New Jersey to sort out the details of locking up the synchronized playback of four 1-in. video machines. Sony was supplying the machines to make this thing work.

Early on, I had decided to use Sony BVH-2000 1-in. video machines for the playback. This decision was based primarily on these machines' ease of synchronizing. With four machines to lock up, this becomes a high priority. There needed to be four machines for security's sake. There were two machines for video, one backing up the other. And there were two machines for digital audio, one backing up the other. BVH-2000 machines include built-in SMPTE code readers and synchronizers. They also include built-in Dolby A cards for decoded playback of the analog audio tracks.

Ahead of the PCM-1610 digital audio processor I placed a video switch that could select between the output of the two machines containing the digitized audio. The powerful error correction of the PCM-1610 provided that this would be a glitch-free switch, should it ever be required. If all else should fail, there was first the analog audio of the first picture machine, then there was still the audio of the backup picture machine. Of course, none of the backup systems were actually needed at the time of broadcast. But if there had been a failure, we had plenty of systems to fall back upon.

Having tested the playback system, we moved all the equipment to the RCA up-link facility at Vernon Valley, New Jersey. This is the largest commercial satellite earth station in North America. Most of the networks' programming goes up to the satellites here. At this installation, not only do they transmit and monitor signals to virtually all channels of both polarities on some seven satellites, but they also measure and control the actual orbital positioning of the satellites.

I made a point of seeing to it that we were able to locate our equipment directly at the uplink. This meant that we eliminated any potential problems from either microwave links or Telco land lines.

We were fortunate to be accessing the Satcom 1R satellite via Westwood One's Scientific Atlanta digital audio distribution system. The system had been used once before for the Linda Ronstadt/Nelson Riddle show, but that had been an analog audio program. This was the first time that a fully digital stereo program was to be commercially distributed by digital satellite.

In the end, we had the PCM-1610 feeding directly into the satellite processor through only a very short piece of cable. It would be difficult to imagine getting a cleaner signal out to the FM affiliates around the country with the current state of the art.

In closing, I'd like to make a couple of observations on the implications of this production. The first is that, with a little planning, the technology is now in place to deliver fully digital audio for video right to the transmitters of local FM stations all around the country.

Also important are the implications for using digital audio as a production tool for the new video release formats. With the new high-performance video formats such as Beta HiFi, VHS HiFi, and video disks finding their ways into the marketplace, their requirements are not going to be met by analog sound tracks. The performance of the release format would excede the production medium! Digital audio for video makes better sense than ever.

But let's not forget that digital audio also makes the best sense for products released in standard video formats. It sounds better in the first place. It edits better. It stores better. And it clones with no signal degradation. The audio for most music videos these days is a good six generations removed from the master. It's no wonder that the broadcasters are complaining about the quality of the audio on music videos. I think this project has abundantly demonstrated that there is no longer any good reason to accept inferior quality audio with video.

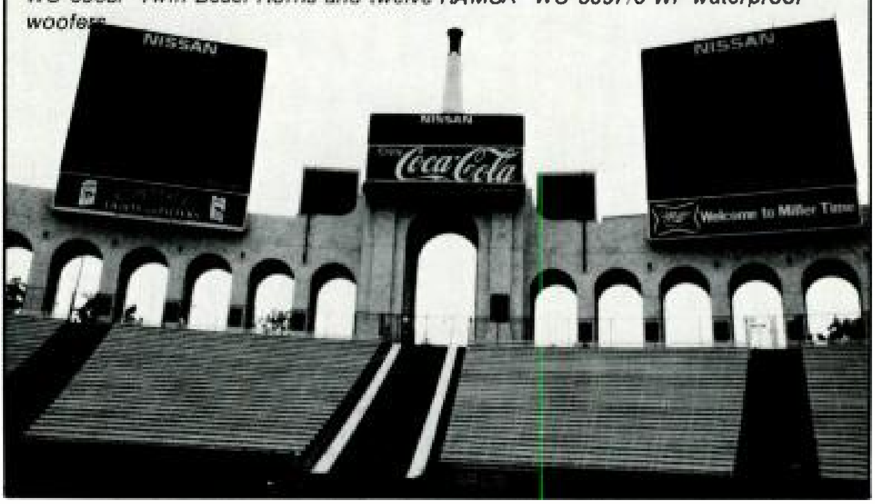
Finally, I would like to thank all the many good people who helped me make this project a success. Unfortunately, they are too numerous to list here. They know who they are. I heartily thank you all! ■

Ramsa At The Olympics

KIMIO TAKEI

RAMSA, a company with the reputation for being a major supplier of console and recording-oriented equipment, was chosen as the sole supplier of audio equipment for the XXIII Olympiad. Read on to see why they were the right choice.

A total of 52 speakers were mounted inside the custom enclosures 15 feet high. In each enclosure are: eight RAMSA WU-5948 Super Directional Horns, six RAMSA WU-5968P Twin Besel Horns and twelve RAMSA WU-5097/8 WP waterproof woofers.



It was probably one of the largest audio installations in history. More than 900 speakers, 250 microphones, 215 power amplifiers, 57 mixing boards and almost 19 miles of cable were orchestrated with exact precision in two dozen locations across a long stretch of Southern California. The athletes were not the only ones who reached new heights at the XXIII Olympiad. Scores of audio engineers whose names will never be recorded in almanacs put their personal and professional best into the Olympic task of making sure the world heard what happened in Los Angeles.

In early 1983, it was obvious to the Los Angeles Olympic Organizing Committee (LAOOC) that there would be plenty of complexities in the sound system without adding to them the problems inherent to any medley of equipment supplied by several manufacturers. The decision was made to work with a single supplier. The supplier would have to be the manufacturer of the whole range of equipment, from mics to mixers, to amps to speakers, and would also have the engineering and scientific expertise to devise sound systems of unprecedented proportions.

RAMSA SELECTED

The committee accepted preliminary proposals from several manufacturers. Obviously it was a prestigious position to be selected as Sound Supplier for the 1984 Summer Olympics. But while manufacturers were concerned with prestige, LAOOC was more concerned that the audio portion of the event be worthy of the athletic excellence it would broadcast.

RAMSA, which is marketed through the Panasonic Industrial Company in the United States, was awarded the contract. Although already having an established reputation in the United States as a major supplier of console and recording-oriented equipment, RAMSA had not yet established its name in this country across the whole spectrum of the sound reinforcement field. In Japan, however, RAMSA has been highly respected for several years for both the sophistication and variety of its products.

RAMSA had already begun to establish a reputation for reliability during large-scale sound reinforcement events. Most impressive was a six-month stint at the 1982 World's Fair. There, 55 WP-9210s, which were to be the principal power amplifiers at the Olympics, as well as some mixing boards, put in 17-hour days, exposed to extremes of heat and humidity as well as an occasional rain shower, without a single failure. Records like that spoke straight to the heart of LAOOC, which had bigger things to worry about than possible power amp failures.

FOUR TRAILERS FULL

The initial proposal and subsequent engineering plans were made under the guidance of RAMSA Chief Systems Engineer Kimio Takei. Smith-Fause, a logically based

---

Kimio Takei is the chief system engineer of the Audio/Video division the Matsushita Communications Industrial Company in Yokahama, Japan.

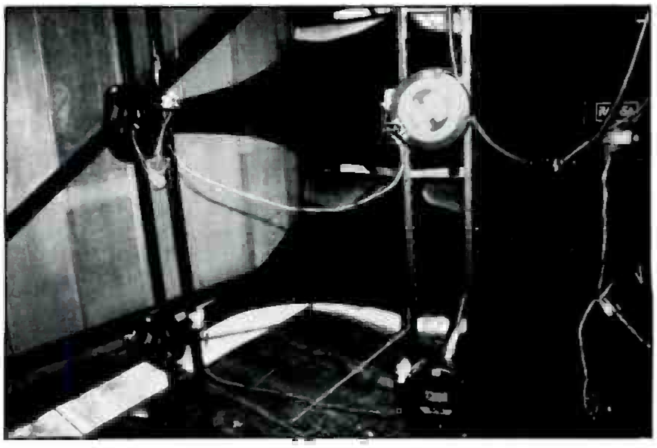

4041
Tektronix
TM 5006
Tektronix
1.0000
.0011
1.00003107

audio consulting firm, contributed to the effort. It was a task calling for considerable knowledge and experience. The venue environments ranged from a 1000 ft. long coliseum, to a lake ¾-mile long to a swimming pool that needed synchronized airborne and underwater sound. Roving sound systems had to be able to play any of several national anthems at dozens of locations. Sound would have to be broadcast various distances—from inside gymnasiums, to the outdoors in stadiums, fields, and even a marina harbor. The resulting equipment list would include thousands of items that would be shipped to venues in four tractor trailers.

Plans had to allow for three separate sound systems at most of the venues. In total there were 20 systems for spectators, 26 for press interviews, 10 for athletic call-up announcements, and 19 portable conference systems. In some cases, the events at a venue required music with certain audio specifications. In most cases, design criteria had to account for the simple reality that the speakers could not stand between the athletes and the spectators or in front of TV cameras or too close for broadcast announcers. Such details had to be coordinated among the architects, builders, television crews, sound engineers and LAOOC committee people.

dB SOUND SUBCONTRACTED

Most installations were temporary. Two, however—at the Los Angeles Memorial Coliseum and East Los Angeles College—were installed as permanent sound systems. In some cases, RAMSA equipment augmented an existing system. In most other cases, a full system was designed and provided by RAMSA.



Inside main enclosures at Los Angeles Memorial Coliseum.

Installation and operation of all systems were assigned to dB Sound, Inc., which is based in Des Plaines, Illinois, and Hawthorne, California. RAMSA chose to subcontract this particular firm not only because of its extensive expertise in high-volume audio installations, but also because of its capability to build custom racks and cabinets. All cabling and assembly fitting was done at the dB shop prior to installation in order that systems could be mounted as quickly and smoothly as possible at the various venues. James Ash, of dB Sound West, oversaw the on-site installations handling whatever changes that had to be made when the physical reality of the equipment had to be fitted into actual venues.

STATE-OF-THE-ART

The spectator sound system at the Coliseum had to be both impressive and technologically advanced. Here RAMSA's equipment and engineering were put to their greatest test. The 25 year-old existing system was considered inadequate by far for a 1000-person choir backed by a full band and no fewer than 50 piano players, all at one end of the 1000 ft. stadium. The new system was expected to perform beyond Olympic proportions.

The primary question at the Coliseum was whether to arrange speakers in a cluster or around the perimeter of the stadium. The following paragraph briefly lists the factors in favor of a cluster design decision.

Under the correct assumption that a perimeter system would have to consist of only horns while a cluster would use woofer speakers as well as horns, RAMSA engineers first assessed sound quality potential. Woofers and a variety of horns would allow good frequency response in low, medium and high ranges, so there would be no compromise on sound quality.

The cluster format also won out in sound clarity. A single sound source eliminated the garbled and unintelligible speech caused by the time difference of sound arriving from several distant points. By comparison, a perimeter system would send out intersecting signals that would become increasingly unintelligible.

Sound pressure level was expected to be about equal, with equal demands on amplifiers and speakers, for both cluster and perimeter formats.

Sound distribution from a cluster source, if properly aimed, can cover the playing field as well as all seats. Perimeter speakers might not properly cover the field.

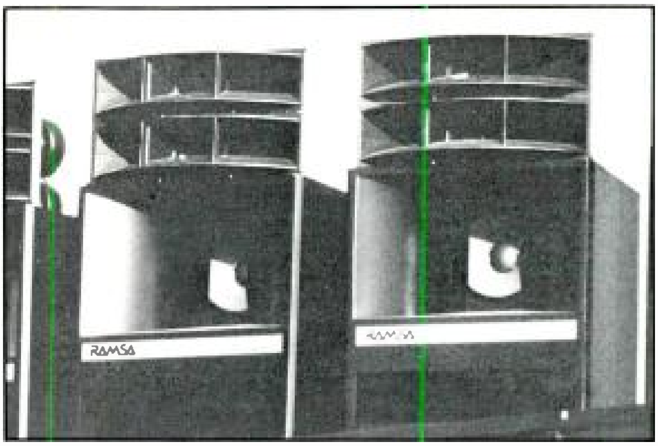
System installation was determined to be easier to facilitate and be more cost effective in a cluster arrangement. No spectator views would be blocked, the sound would be coming from behind, and it would not be necessary to erect numerous poles on which to mount numerous horns. The cost of low and high frequency speakers around the perimeter would have a factor, as would the cost of wiring the entire coliseum. Maintenance would also be more difficult for a perimeter system.

A cluster system would also limit power transmission loss because power amplifiers could be located near all the speakers. Low voltage, low impedance on heavy gauge wire would result in good overall system efficiency.

91 dB AT 1,000 FT.

RAMSA determined that a cluster system was best suited for the application. A pair of clustered speaker arrangements would sit on both sides of the peristyle, which held the Olympic torch. Coincidently, the clusters would stand beside the giant video screen that was supplied by Panasonic Industrial Company. A total of 52 speakers were mounted inside two custom housings 15 feet high. In each were eight RAMSA WU-S948P Super Directional Horns, which could project sound over 1,000 feet; six RAMSA WU-S968P Twin Besel Horns; and twelve RAMSA WU-S907/8WP waterproof woofers.

Proper alignment of this array was no simple matter. The goal was 91 dB at the opposite end of the stadium, enough to overcome an ambient 85 dB of crowd noise. The sound had to be evenly distributed throughout the coliseum, including the field, while not overpowering spectators nearest the speakers. This was possible since the low frequencies of the woofers would spread evenly while the medium and higher frequencies could be aimed precisely at the various target segments of the stadium.



Large RAMSA enclosures sit atop Pepperdine University swimming stadium for the water polo events.

As long as the horns were aimed and regulated properly, they would be able to spread an even mix of frequencies to all points.

FROM DRAWING BOARD TO SCOREBOARD

That was the concept. But how to bring it to reality? Given a few months of practice time, it might have been possible to align the 52 speakers in a variety of permutations, set and reset the mixing console and run up and down and around the bleachers until it sounded just right everywhere.

Instead, RAMSA utilized the resources of its Central Research Lab in Osaka, Japan. There, a VAX 11/780 computer that is programmed for acoustic ray simulation techniques provided graphic readouts of individual speaker and clustered speaker acoustic coverages. Contours emanating from a cluster point at the head of a computer model of the Coliseum clearly and graphically indicated how the aiming of each speaker affected overall coverage. The rack on which the speakers were mounted had degree compasses to allow each speaker to be aimed precisely as the computer dictated.

RAMSA installed 30 dual 200W WP-9210 amplifiers to power this array. To insure a properly balanced sound the amplifiers were used in combination with WZ-9320 graphic equalizers and WZ-9420 crossovers. Coordinating the complex input and output signals was a RAMSA WR-8616 mixing console. This console, which had already proven itself in broadcast, recording and postproduction assignments, successfully handled both the spectator sound system and the broadcast feeds to ABC-TV. The WR-8616's balanced input and output design provided high quality audio signals for both the broadcast sound and spectator sound requirements.

Also at the LAOOC headquarters was an equalizer recording room where cassettes and endless loop cartridges were high-speed duplicated for use at the various venues and with the roving sound systems for the award ceremonies. Technics dual deck cassette recorders were used as were Technics turntables and monitor speakers. National anthems, played for Gold Medal winners, were on endless loop cartridges and in most cases were carted from winner's stand to winner's stand as needed.

The three press interview systems at the Coliseum were entirely separate from the spectator public address system. Like all press systems at the venues, these

systems each used a RAMSA WR-130 mixer, a RAMSA WP-9210 power amp, several RAMSA WM-8080P mics, a pair of RAMSA WS-100WP speakers, and a RAMSA WS-EQ1P graphic equalizer. The equalizer was used to selectively reduce low frequency sounds to avoid the feedback caused by the microphones being so close to the speakers. All the equipment could be easily packed into a rolling case for transporting.

The sound control equipment and the operator were housed in a second story room in the press box. Technics speakers monitored output. Input and output levels were set during a full rehearsal two days before the opening. Athletes, choir, band and all participants went through their motions before 40,000 spectators. On the day of the opening, the operator had little to do but make scheduled pre-set adjustments.

6 MILES OF CABLE

The Los Angeles Coliseum was just one of a dozen venues with a full sound system. While it may have been the most technologically challenging, it was not the most physically demanding. That distinction belongs to the steeple-chase endurance course at Fairbanks Country Club. There, for a one-day event, *dB Sound* installed over 60 RAMSA WT-900P horn speakers. Most of them were set atop 15-ft. poles and aimed down at each of about 30 spots where spectators congregated to watch the competitors tackle a particular obstacle. Some six miles of cable was laid to connect the speakers to the centralized WR-130 mixer, four WP-9210 power amps and four RAMSA speaker transformers. Musical accompaniment was played through a Technics RS-M234 cassette deck.

A separate athlete call-up system was also installed in the stable area. It used a WA-750P mixer amp, WM-8080F mic, and 13 WS-1700P bi-directional speakers.

SOUND BY THE SEA

Another widely spread-out venue, the Long Beach yachting area, was installed with a system similar to the one at Fairbanks. A WR-130 mixer and a line transformer fed three WP-9210 power amps, each of which fed twenty WT-900P horn speakers. For ease of operation, the mixer and amps were centrally located, though this meant laying some four miles of cable. For the speakers located at the end of five 200-ft. piers, cable had to be draped through the water and weighted to the bottom by cement blocks.

This multi-purpose system was used to make announcements to spectators, call up athletes, and play background music.

UNDERWATER SOUND SYNCHRONIZED

One of the most technically sophisticated systems was installed at the U.S.C. swim stadium for the synchronized swim event. The swimmers were not the only ones who had to be synchronized. The music to which they swam came from a cluster of two WU-S948P single direction horns, two WU-S968P horns and four WU-S566/8P drivers. Another set of special speakers generated sound under water. The problem was that the airborne sound from the cluster would arrive at the pool some time after the wire-borne signals arrived at the underwater speakers. So unless there was time compensation, as the swimmers' routine passed from above to below water, they would be subjected to a break in music tempo.

To overcome this difference, a RAMSA JJ-193 digital delay was installed. The underwater sound was then in time with the above water sound as it reached the pool. An underwater pick-up/transducer fed the subaqueous sound to a headphone so the system operator could verify successful simultaneous transmission.

Other equipment here included two WR-130 mixers, two WA-750P mixer amps, seven WP-9210 power amps, and a WZ-9420P crossover. A Technics RS-245 cassette deck provided the music.

The mixer accepted input from 3 WM-8080P mics, the cassette deck and the underwater pick-up. It fed a 1 x 16 mult-box that could feed up to 16 press tape recorders. This mixer also fed a mixer amp that supplied the cluster, and another mixer amp that fed the underwater speakers and the operator's headphones.

STADIUM STRATEGY

Sound systems for other stadiums were all similar cluster speakers. The Vellodrome, which held cycling events, used a WR-130 mixer to feed eight WP-9210 power amps. Twelve WU-S968P twin besel horns, twelve WU-S566/8P drivers and eight WV-S907/8P woofers were divided into two clusters that were aimed at the grandstands on either side of the track.

At East L.A. College, for the field hockey games, two clusters were built of two WU-S948P super directional horns, ten WU-S566/8P drivers, eight WU-S968P twin besel horns, and twelve WU-S907/8WP waterproof speakers. The system used a WR-8616 mixing console, nine WP-9210 power amps, two WZ-9420P crossovers and two WZ-9320 graphics equalizers. This system was installed as a permanent addition to the stadium.

At Santa Anita Racetrack, where the equestrian events were held, a single large cluster of speakers hung over one of four surrounding grand stands. The speakers were aimed, with varying connections, around a full circle. Supplemental horns, fed through a digital delay to match the sound timing, were installed at the far grandstand to give the entire spectrum area clear and adequate sound. A WR-130 mixer and twelve WP-9210 power amps fed four WU-S948P single direction horns, eight WU-S968P twin besel horns, twelve WU-S566/8P drivers, and twelve WU-S907/8P woofers. A Technics RSM-234X cassette deck supplied music. The athlete call-up system at Santa Anita was composed of two WA-750 mixer amps feeding sixteen WT-900P horn speakers that were spread throughout the stable area.

At Pepperdine University, for the water polo event, a single cluster was aimed in three directions to hit the three grandstands and pool. A RAMSA mixer, amp and crossover fed eight divers, eight WU-S990P radial horns, and four WU-S507P/K3 woofers.

The archery venue at El Dorado Park was extensive but relatively simple. A mixer, amp and speaker transformer fed nineteen WT-900P horns which are aimed facing the audience. Four other WT-900P horns powered by a WA-750P mixer amp, were used for athlete call-up.

Similarly, the boating events at Lake Casitas used a line of over fifty horns to face an audience that was spread across ¾-mile of shoreline. At a central location there was a cluster of four WU-S566/8P drivers, four WU-S990P radial horns, and two WU-S507/K3 woofers to provide sound reinforcement for the Dixieland Jazz bands entertaining the crowds before, after and between events.

The RAMSA sound system did more than provide exceptional sound coverage. It firmly establishes RAMSA's presence and potential in North America. ■

Renovations At Montreal Sound

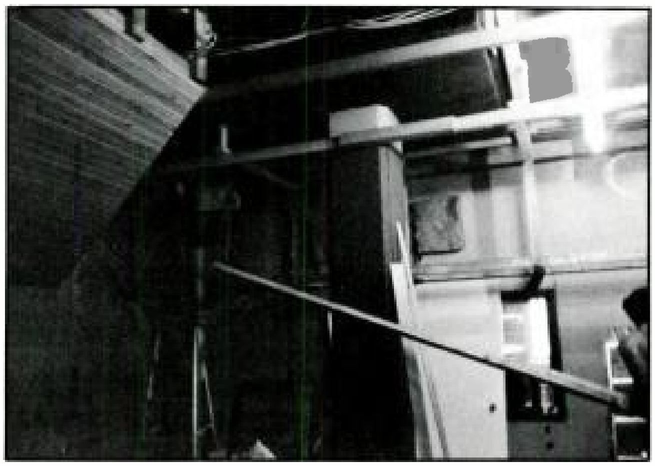
ROBERT BREWSTER

Well, we're taking off again to the Great White North! This time to get the following account of the good, bad, and the ugly aspects of renovating Montreal Sound Studios.

WELL, HERE WE GO AGAIN! Who would have figured that the folks at db would let us crazy Canucks run rampant through the pages of another issue. You know, eh, that by the time you read this the first Canadian will have taken a ride aboard the N.A.S.A. Shuttlecraft? Though anyone who read the May '83 issue of db knows full well that the boys at Montreal Sound recording studio have been in space for years. After that article appeared, we were all convinced that we were legends in our own minds. In fact, we sat around and waited for Johnny Carson to call; he didn't. Neither did Merv or Mike. Can you believe that? Not even Alan Thicke called. Since that time, a lot has happened around Montreal Sound. I'll tell this story in the style that America loves best: those three unforgettable Clint Eastwood westerns. First—the GOOD, the BAD, and the UGLY.

THE GOOD

Back in the fall of 1982, after the completion of our fancy acoustically designed and constructed control room, you might say things in the studio business took off faster than a Wayne Gretsky slapshot. Word spread very quickly that state-of-the-art sound was now available in Montreal at a very affordable price. Some producers even swore that our finished masters had this magical ingredient that other studios at twice the price didn't. To put it in basic American, "We had the beef." Our new control room was the talk of the town. If Bill had wanted to, he could have booked the studio twenty four hours a day. As for myself, I secured a few TV voice-overs for Canada's major airline company and my favorite beer.



Bob Maher and crew renovating the ceiling and air conditioning system.

THE BAD

The economic recession of "82-83" hit Canada harder than the U.S., which meant that our jingle and radio commercial business almost went out the window as retailers of all sizes were deciding to hang onto their advertising dollars. Because of the uncertain times, three of the stars of our control room construction were not around for the latest insane project. Unwillingly, they moved on to more stable careers.

Because of the increased business at Montreal Sound, Bill was finding that he spent more and more time with the tedious but very important day to day operations of the studio, plus he was still engineering most of the sessions. All of you in the studio business know all too well how superstitious and finicky record producers and musicians can be. Since Bill engineered their last record and found the hit groove they were looking for, they insisted he be behind the controls for the next one.

One client even went as far as saying, "Only Sweet William can capture the cosmic vibrations and reverberations of my reggae music and transform them after a little *edit-size-ation* into a glorious sanctified revelation." I'm not sure, but I think that meant he'd only work with Bill (Sweet William) Hill. Even the ad agency clients who are, as a rule, more reasonable and generally not too superstitious, were insisting on doing all their sessions with Bill. Even Daniel Seguin, the designer and our "Mr. Technic" around the studio was suddenly in great demand himself. He was being called away for weeks at a time to perform his magic for other studios. This meant that Bill was working day and night, and quickly going out of his mind. Besides me, who is not too adept at anything much else than standing in front of a microphone or sitting in front of a typewriter, or maybe answering the odd phone call or two, here was poor Sweet William having the best year in the studio's history and doing it all by himself.

THE UGLY

All those hit records that Bill and I were going to write and record were put on the back burner. Everyone wanted the studio, but only if it came with Bill. Being a true Scotsman, he couldn't say no. Then it happened. For months Bill had been talking about doing lots of crazy things, but finally he sounded like he really meant it.

Robert Brewster, Jr. is a freelance creative writer, specializing in radio and television jingles.


Finished live area with curtain open.

Bill: This is it. Today I'm finally going to resign as Studio Manager, and even give up engineering completely.

Bob: Uh huh, then what?

Bill: Next week, me and the family are moving to New Jersey where I can take up surfing and sell sea shells to the tourists.

Bob: Sounds lovely, Bill, wish I could join yuh.

The ugliest thing of all: Bill's family was so deprived and desperate for a little attention and fatherly love and they became so disillusioned and confused that they actually encouraged him. Now most people would pity a man in a situation like this, but not me. After all, it was his own fault. The man had become too good at his job.

Well, just when it looked like the sun was about to fade on Sweet William and for sure it was, I heard something. Three seconds before Bill's main fuse was about to blow and overload all over the place, far off in the distance the faint but very distinct sound of the cavalry coming to the rescue was riding in on the North Wind. As Bill reached up to tear out what little hair he had left, our would-be hero zoomed up on his charger.

A FISTFUL OF DOLLARS

From out of the north came this screech of tires and a bright red Porsch. The door flew open and out stepped Ken Hallam. Ken was a Canadian oil man. He's a man who has dealt in bottom lines as a way of life and has been very successful at it. Here was a man who, on the surface anyway, appeared to have all of his vital senses intact.

To get to the bottom line of all this, Ken was, in fact, just like all the rest of us in the business of making music—crazy. Earlier this year, he had bought into Montreal Sound as a working partner. He took over the business operations of the studio, and ran the newly formed record company that wanted to see if Bill and Bob could actually write and record that hit record they kept talking about. I still wasn't thoroughly convinced that this was going to work until the day Ken walked into work followed by a man pushing a telex machine.

Daniel: What the hell is that? A new typewriter?

Bob: No, it's a telex.

Daniel: Telex, smellex, who cares? We don't need that.

Then Ken looked Daniel straight in the eye, smiled, whipped out his checkbook and said, "How much?"

Daniel: For everything?

Ken: Everything.

Daniel: About $800.

Ken wrote the check. Daniel, meanwhile, was dazed, fazed, in a haze, and his mind in a state of total disbelief. You see, he'd been bugging Bill for months now to take care of this stuff, and was beginning to believe that he never would. But with one swift stroke of a pen, Ken Hallam was restoring Daniel's faith in mankind and life in general. Anyway, it wasn't long before the word spread all the way up north to the home and workshop of Bob Maher, our company carpenter at large. Then there was a new influx of capital at Montreal Sound. Those of you who read our control room story know why we affectionately refer to Maher as "Nightmaher." The rest of you can read on. It was June 15 when Maher came face to face with the man who held the fistful of dollars. Poor Ken, not realizing that Nightmaher was not your average run-of-the-mill master carpenter, but rather an artist who dreamed in panavision and quadraphonic sound, got drawn into the dream, and the fistful became smaller. Daniel, knowing a good thing when he saw it, conspired with Maher and came up with the perfect plan for a new fancy acoustically designed studio, sound proof office, and lounge areas.

Daniel: While we're at it, we should have all the strips in the Harrison console tuned up in the lab as well as replace the diaphragms and drivers in the monitors. Plus we'll buy a new Studer 810.

Maher: I'll tear everything down except the control room. There will also be oak everywhere the eye can see.

Daniel: I'd like to rewire the whole studio and relocate the amp rack to another room. Maybe I'll design something to accommodate it.

Maher: Do that. I'll build a cedar box around the rack and put in a matching hard wood floor and brand new doors everywhere.

Daniel: We should also install a silent air cooling and ventilation system, running through the whole complex with vari-speed adjustable ION controller.

Maher and Seguin both broke into hearty laughter, as they realized that even they were getting more than a little carried away, or were they? The boys' laughter was reaching hysterical proportions as they took turns blurting out other great insane ideas for the studio. Suddenly, the office door flew open and Ken Hallam stood in the archway.

Ken: What's so funny?

Well, what happened next went something like this. Maher and Seguin jokingly told Ken about their elaborate, extensive, innovative, creative, impressive and very expensive plans they had for the studio.

Maher: Yeah, Ken, the best thing about it, you'll only have to sell your Porsche to pay for it.

Daniel: Yeah, and maybe your golf clubs too.

It seems that the man takes his car and his golf very seriously. And, well, Ken was staring Maher and Seguin up and down and pacing around the room. Then like a flash, Ken's checkbook flew out of his pocket.

Ken: Okay, boys, let's do it. You'll start the first of next month and it better turn out looking and sounding as good as you say, or else.

Well, that fistful of dollars got a little smaller still and for the first time in many years Ken was facing the prospect of seeing the palm of his hand. As you can tell, it was worth it.

The whole job took about five weeks to complete and was constructed by Bob Maher and a crew of four hand picked men. Maher learned his lesson the last time, by relying on studio people as helpers. All this meant that Bill was off on his first real vacation in eight years. That's how long Montreal Sound has been in operation.

It should be mentioned at this time, that a few weeks after Ken arrived on the scene so did James "The Kid" Carrier. The Kid was, what we like to call in the Canadian Studio business our "resident in-house all round studio person in training." In other words, he'd be doing all the real dirty work no one else wanted to. The kid, like the rest of us, was crazy. He had to be to continually take the non-stop abuse for weeks on end. Why? Because he knew he was being groomed to sit in Bill's engineer chair. And if

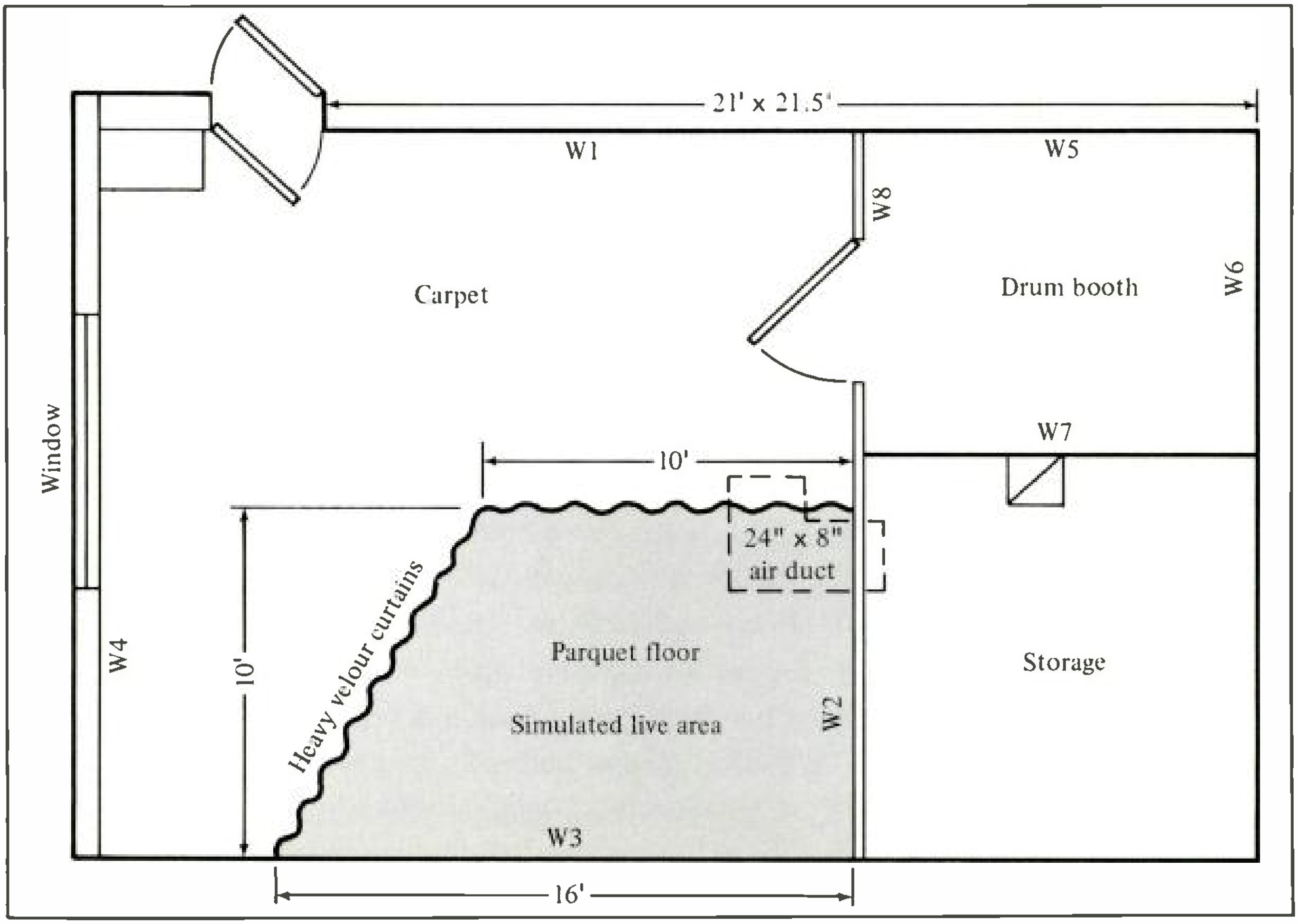


Figure 1. Diagram of floor plans of studio area.

everything worked out he would become the permanent heir apparent in waiting. Though we never talked about it, we all knew that one day the old man would have to hang up those headphones for good and retire to a quiet life of... whatever...

THE LAST DAY...

This scene took place on the last official work day. In a way, it sums up how Montreal Sound's latest adventure into the world of renovation and expansion affected the people involved.

Bill: Somebody pinch me quick. Is it really real? Am I standing in the studio I always dreamed of?

Ken: Yeah, it's really real, and I'm finally in the music business.

Maher: So, Ken, what do you think?

Ken: It's everything you said it would be, plus a lot more.

Daniel: You can thank me for that. It's all because of my great acoustically designed plans.

Maher: The best laid plans ain't worth anything without a great artist to bring it all to life and that's me. Right Ken?

Ken wasn't really paying attention to what Maher was saying. He was too busy adding up the total expenditures for the complete job.

Maher: You know Ken, for a few dollars more this place could be perfect. Can't you just picture some lovely oak beams running the whole length of the studio?

The only thing Ken Hallam heard Maher say was, "for a few dollars more," and since he just looked at the grand total cost of the job, Nightmaher was in big trouble. Bob Maher turned around just in time to see Ken in the middle of his windup.

Ken: Please Maher don't move. This is gonna make my day.

Maher: Okay, okay. Let's forget all about it and call the job a rap.

THE FINALE

Ken felt quite good as he looked around and surveyed the whole place. We all felt good. the studio looked fantastic. and better than that. the sound was out of this world. I'm sure that we at least doubled our capacity for cosmic vibrations, and tripled our sanctified revelation potential output. I know this for a fact because Daniel told me so. You see, he had a special computer program written up to calculate the new studio room volume and reverb time at each frequency octave band from 125 Hz to 4 kHz. The program took into account all coefficients of absorptions and all surfaces of material in square feet or square meters. With this information, it was very easy to find the room reverb time by changing materials to different surfaces. All the walls in the studio are subdivided into one, two, or three surfaces. The desired result was obtained by assigning the proper material to each of the surfaces. The live section of the room has a reverb time of more than ".85" seconds taking into account only the reflective surfaces; while the others are considered free field. Curtains were installed in the room that could be opened and closed. When shut, the room has a fairly dead sound with a reverb time of less than ".8" seconds. I've been told by Daniel that we now have an endless amount of ways to alter the acoustics of the room. Also, because of the computer program. our new studio is as close to a perfect acoustic match to the control room as is humanly possible.

Well, folks, that's about the whole picture. Until next time, this is America's Canadian eye saying. "Happy trails to you, eh!" ■

Notes From The Resident Technician

In relation to the technical updates at Montreal Sound for calculating the reverb time of any room, Dan Seguin developed a computer program in BASIC that gives a graphical representation. This program computes room modes for resonance problems, room volume (different sub-routines for different room shapes) and reverb time curves and tables.

The program DATA includes:

1. 25 surface area values (square feet or square meters).
2. A matrix of all materials and the corresponding coefficient of absorption at: 125 Hz, 250 Hz, 500 Hz, 1 kHz, 2 kHz, and 4 kHz.
3. Room dimensions.
4. Room shape codes.

The room floor, ceiling, and walls are divided into 25 surface values. Each surface represents a selected material. In some other cases each surface could represent different materials. But the main purpose of this is to be able to select a material for each surface by pressing only a few keys on the computer and getting a reverb time curve in no time. (Each of the surfaces on the plans carry a number: *S* "*X*." It is easy to repeat this process as many times as we need and get a preferred T_{60} for a particular room.

The materials used for Montreal Sound are simple, inexpensive, and cost efficient. They include: Pine, cedar, carpet, heavy velour drapes, glass, mineral, iso-fiber, and 2-in. Sonex. Using a floppy disk, curves for each room in different conditions can be filled. (Different conditions apply for a room with variable acoustics.)

At Montreal Sound's main studio we have variable acoustics because the heavy velour curtains can be closed, partially closed, or simply left open. The room becomes more absorptive when curtains are closed, and also this eliminates the first few reflections (flaps) at the live section walls, which in some cases is not desirable. At the same time the live area becomes isolated to some degree. The purpose of this is to give us some flexibility for recording...

Equipment List

Console
Harrison 4032B
Automation
Allison
Studio Monitors
JBL 4311
CTR-Room Mon
Westlake TM1 (JBL 2235H Woofers)
Amps
1 Yamaha P2050
2 Bryston 4B (bi-amped)
1 Crown DC-300
24TK Machine
MCI JH24
2TK Machines
1 Studer B67
1 Studer A-810
Comp-Lim/Gate
UREI LA3A
Symetrics 522
Melcor
Kepex Gates
Delay
Lexicon Prime Time
Reverb
EMT 262 plate reverb
AMS RMX-16 digital reverb
Yamaha R-1000 digital reverb
Effects
Flanger Eventide
Harmonizer Eventide
Delta-Lab Delay/flanger
Mics
Many kinds of the most popular mics
Instruments
1 Yamaha 9′ Grand piano
1 DX7 Yamaha
1 Prophet 600 Sequential circuits
1 Drumulator
1 Apple II Computer with Roland compu-music program and interfaces
Cassette Decks
Sony & Nakamichi
Headphones
AKG 240
Headphones Distribution Units
Dan Systems
Console Direct Input Interface Unit (located in ctr-room)
Dan Systems

Equipment Renovation List

1. All capacitors in Harrison console have been replaced.
2. All EQ frequency selection pots have been replaced in the console.
3. One ctr-room equipment/effects rack has been installed on an elevated rotation frame for easy maintenance access.
4. One new air conditioning unit was installed for the control room.
5. New parquet floor behind console for engineer chair flexibility.
6. Relocation of all amps into a special high ventilation back access rack. Located outside control room in a separate room for that purpose. Air filters and powerful fans provide clean air for cooling.
7. One front glass door for access to controls.
8. One side door for maintenance.
9. Renovation of all offices, rest areas, etc.
10. Installation of one additional 100 amp, AC powered braker panel for extra lights, outlets, air conditioning, etc.
11. Renovation of main studio, adding a live area, new acoustics and lighting (including the drum booth).

Surface Values: (sq. ft.)

S1 = 124.5
S2 = 50.75
S3 = 24
S4 = 22.75
S5 = 62
S6 = 105
S7 = 10
S8 = 9
S9 = 9
S10 = 18
S11 = 12
S12 = 10.5
S13 = 136,5
S14 = 129.5
S15 = 26.5
S16 = 3
S17 = 10
S18 = 113.75
S19 = 0
S20 = 26.75
S21 = 70
S22 = 17.5
S23 = 300.5
S24 = 130
S25 = 300.5

VOLUME VALUES: (cu. ft.)

Drum Boots	512
Main Boom Curtain Open	4861
Main Boom Curtain Closed	3741
Live Area Curtain Closed	1120

TABLE 1: SABINS T_{60} (secs)

(kHz)	.125	.250	.5	1	2	4
Drum Booth	.1	.17	.12	.13	.14	.13
Main Room (Curtain Open)	.42	.4	.31	.32	.33	.31
Main Room (Curtain Closed)	.3	.27	.21	.21	.21	.20
Live Area (Curtain Closed)	.6	.5	.41	.37	.37	.4
Live Area (Curtain Open)	.8	1.06	1.25	1.88	1.8	1.8

(assuming other surfaces = free field)

Sound Reinforcement In Asia

Engineer Ed Learned recounts details of his six week tour of Asia as sound system designer and operator for the Chico Freeman Quintet. Come along on his journey and get the inside story on sound engineering abroad.

IT'S A "GIVEN" that when you work in professional sound reinforcement, you are going to travel. In over 10 years in the business, I've seen most of the U.S. and Canada. I dreamed, however, of someday seeing the Taj Mahal, Pyramids, and the Great Wall of China. Little did I realize that when I picked up the phone in the fall of 1982, I would soon realize some of these dreams. A personal recommendation resulted in my being hired as an engineer for the Chico Freeman Quintet. Chico and his band were to embark on a six week tour of South Asia in the winter of 1983. From February 14 through March 28, 1983, we were to visit Quatar, Pakistan, India, and Sri Lanka. My duties on this trip were to function as sound system designer and operator.

PUSHING THE LIMIT

Sound equipment must meet several important requirements for use in Asia. Concerts are held anywhere from an ambassador's living room to a 3,000 seat open air amphitheater. Each piece of gear must be able to fit through a small jet's cargo door, and the total system weight cannot exceed 1500 pounds. Considering the wide variety of venues, flexibility becomes paramount. All different types of power and coverage requirements exist: low power, wide dispersion for the concert on the courtyard, but long throw, high power is needed for the 2000 seat theater. Sound reinforcement for jazz must also meet more stringent standards for fidelity, with little or no coloration from the speaker system. *Full-range* type cabinets can provide controllable coverage in a small space, so I favored these designs as an overall system approach. Front-loaded type enclosures were preferred over cone speakers to avoid coloration. Electricity is 220-240 volt, 50 cycle AC, so a transformer is needed to "drop" the voltage to U.S. 110-120 volt operating limits. Because power surges and drops are more common in Asia, it's important to consider components that can easily tolerate voltage fluctuations.

Equipment does not travel by air-ride semi-trailer in Asia, it travels over bone-jarring roads and air freight handlers, baggage porters, and stagehands all contribute their special abuse. All equipment must be securely packed and protected. Simplifying system components is an important part of durability. Limiters, effects devices, and electronic crossovers could all disable the system by breaking down. A system with good passive crossovers could actually be advantageous, not to mention the weight saved in multiple amps.

AERIAL ENTERPRISES COMES THROUGH

I decided to use Aerial Enterprises, Inc. to provide the PA system. Aerial, founded in 1976, is well known for their high fidelity speaker systems. I had done tours with the company before, handling groups as varied as the Spinners, John Cougar, and Grand Funk, so I knew the gear could hold up. But it is the overall Aerial design philosophy that makes its equipment especially well suited to an Asian tour. All Aerial's speaker cabinets are based on front-loaded, ported, and ducted enclosures using Gauss loudspeakers. Horns and compression

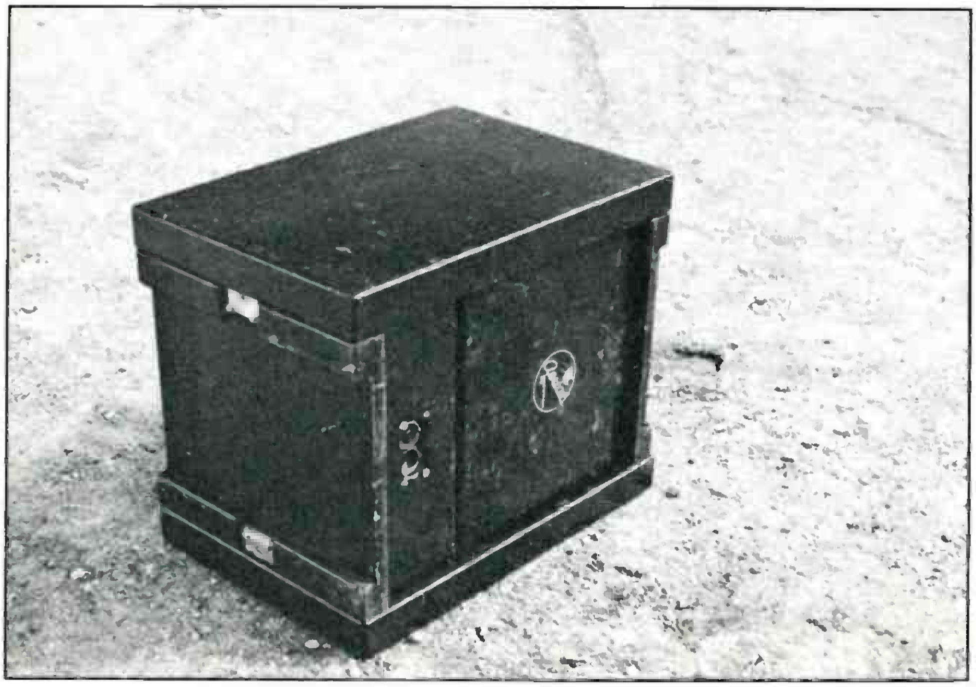
---

Ed Learned has been working in professional sound reinforcement for ten years and is now a chief engineer at Aerial Enterprises, Inc.

One of Aerial Enterprise's monitor enclosures with Gauss loudspeakers.

drivers are used, but are typically crossed over around 2000 Hz to allow cone speakers to reproduce more of the harsh midrange area. Tweeters are also employed for full treble response. "One-piece" cabinet design and narrow-throw horns help enhance Aerial's "controlled dispersion" concept. Cabinets can be used to smoothly cover a well defined area. With proper stacking techniques, this "arc of coverage" can be expanded to meet changing venue coverage situations; just what is required in Asia.

Aerial has many cabinet designs; "full-range" boxes as well as component stacks, all based around 18-in., 15-in., and 12-in. Gauss loudspeakers. All dual driver cabinets are time aligned, while the depth and rail structure assure time alignment between cabinets. Due to the size and weight requirements, the best sound in the smallest space proved to be the Aerial floor monitor. This is a front loaded, ported and ducted enclosure loaded with a Gauss 2842 12-in. loudspeaker and an Electro-Voice T350 tweeter. A passive crossover is employed that operates at 5000 Hz, 18 dB per octave, and pads the tweeter down to compensate for its greater efficiency over the loudspeaker. There is also an EV tweeter protector included. The cabinet is internally braced, not only at the corners and edges, but on the faces as well. The entire cabinet is one piece, with no removable front or back. Covered with a black epoxy finish, the cabinets are not only durable, but are so airtight they float! I was somewhat skeptical that a floor monitor with a 12-in. woofer and tweeter would have the necessary fidelity, but testing with a Crown RTA-2 revealed a frequency response of 40 Hz to



Monitors with packing trays attached for shipping.

22 kHz, overall and 60 Hz to 14 kHz ± 3 dB A-weighted. Mirror imaging allows for coverage variations of varying types (see photo 1). Because the same type of speaker cabinet is used for house and monitors, everything is a spare for all uses. The connectors are Hubbell Twist-Locks, recessed to prevent damage. For packaging, two monitors are placed upright back to back to form a rectangle, then placed in wooden trays. End pieces are placed over the speaker grills which fasten to the wooden trays with roto latches. This forms a wooden exo-skeleton that is braced internally by the monitors themselves.

For this tour I used 2 Crown DC-300A series II amps, mounted in an Aerial DCA rack. The rack totally encloses both amps, with a removable front panel for amp access. Two fans positioned in the center of the rack provide filtered forced air cooling off the amps. Inputs are transformer isolated to eliminate grounding problems, and use male and female XLR type connectors. These incorporate built-in Y connections, so the same signal can be jumped from input to input, or to another rack. Both the XLR inputs and Hubbell Twist Lock outputs are recessed to prevent damage. To provide support, the amplifiers' power transformers are supported top and bottom by wood blocks that are permanently attached to the walls of the rack. This helps support the amps' weight, and relieves stress on the faceplates. This rack then fits into an overcase that completely surrounds the rack with 3-in. of foam on all sides for shock mounting.

The console for the tour was a 14 input XPC-2, manufactured by Custom Audio Electronics. This is a small, straight-to-stereo console that can be any size from 1 to 38 inputs. The modules stack and attach sideways to each other—no mainframe! Each input includes continuously variable gain (40 db), two switchable monitor sends, bass and treble shelving equalization with switchable frequencies, fader, pan pot, and stereo master assign switch. There is a pre-fade post EQ solo switch for headphone monitoring, as well as an LED level meter, reading -20, 0, and + 20 dbm input levels post EQ.

In a small electronics case I carried 2 EV-Tapco 2210 stereo graphic equalizers, to provide equalization for house P.A. and 2 monitor mixes. This gave me a spare channel of EQ as a backup, or to equalize a P.A. augment should it be necessary. To round out the house electronics, I carried a VIZ 120B power line monitor, a peak-reading AC voltage monitor. This enabled me to monitor fluctuations in AC power during performances, and adjust my transformer output accordingly.

The transformer was a 28 amp Variac multiple-tap, with an output voltage of 0-280 volts. Fifty or sixty cycle current can be used with this unit. Shock mounted with three inches of rubber and wooden bracing, the transformer had standard U.S. 3-pin grounded outlets and a Hubbell Twist-Lock inlet. The AC feeder terminated into three tails: hot, neutral, and chassis ground. This could be connected to building mains, or attached to the AC connector appropriate to the country being visited. I included some extra wire, enabling me to run my own chassis ground to a cold water pipe or other grounding source, as it is not uncommon to find facilities with no equipment grounding in Asia.

All speaker cables were 14 gauge with Hubbell Twist-Lock connectors. Nineteen conductor 125-foot and eleven conductor 30-foot snakes served as main and stage snakes respectively. Mic stands were Atlas straight stands of varying heights and Beyer boom tops. For microphones, I used EV DS-35s, AKG D-200Es, and Crown PZMs (31-S). PZMs are especially desirable on grand pianos: They sound great, and they save on mic stands! I also carried an Aerial direct box for use with the string bass. These boxes operate like a standard D.I., but also include a padded input (-30 dB) so the box can be used post-amplifier, on a speaker output.

The total system of eight floor monitors, amps, mixing board, transformer, and cords/cables consisted of twelve pieces, with a total system weight of 1450 lbs. Four monitors were used for house PA, stood upright and stacked 12-in. to 12-in. for increased bass response. The remaining four monitors were used for two monitor mixes, two per mix. Total house PA power was slightly more than 600 watts, each monitor mix around 300 watts.

I planned on taking my Anvil briefcase along with me as carry-on baggage, so it became the system "toolbox." Besides my tool roll, soldering iron and volt-ohm meter, I carried extra mic clips, fuses for everything, fuse caps, audio adapters, ground lifters, and assorted detritus. I even tossed in some odd power supply parts for the console as an afterthought. Consider Murphy's Law; it's always best to be prepared for the worst.

AND WE'RE OFF

After arranging to ship the system air freight via London to Quatar, I drove to New York to meet with Chico and the quintet before leaving the U.S. The group consisted of Kenny Barron (piano), Wallace Roney (trumpet), Clarence Seay (bass), and Ronnie Burrage (drums). Chico played tenor and soprano saxes, bass clarinet, flute, and bass flute.

Our first meeting, on Saturday, February 12th, was to meet with the USIA agency escort, Ivan (Ike) Izenberg, who would be accompanying us on our tour. Ike was to brief us on the travel schedules, performance sites, and any other details pertinent to touring these parts of the world. This was during the famous February snowstorm of 1983 and there was at least 1½-feet of snow in midtown Manhattan, with nasty drifts. Nothing moved, except for a few subways. Despite this, the briefing was held with a little help from *Ma Bell*. Afterwards, Ike and I attempted to determine how the weather would affect shipping our air freight. All airports were closed, but the Pan Am freight supervisor assured us (cross your fingers) that our shipping deadline would be met. At that point I could relax a bit and enjoy the rest of my weekend in the Apple.

We all rendez-voused again Monday night at JFK airport (now open) and, after the inevitable delay, took off for London. We arrived the next morning, transferring to Gulf Air for our flight to Doha, Quatar. We arrived late in the evening and, were whisked through customs and to our hotel, where we were to play two concerts on successive evenings. Only two problems—no baggage and no sound equipment! Neither had made their respective flights. A shopping trip the following day helped ease the temporary loss of personal effects, but the sound gear could not be replaced so easily. At a reception for us that evening, I was notified that the sound gear (and our bags) would be arriving the morning of the 17th, and delivered to the hotel ballroom in time for a noon stage call. The sound gear had been sent to Miami, Paris, then Doha—a rather roundabout route.

Sure enough, in the morning everything arrived. I set up and tested the PA and the band came in around 1:00 PM for a three hour rehearsal/sound check. The stage plot illustrates stage positioning and mic placement, although we did experiment a bit. Keep in

mind that we carried drums, horns, an acoustic bass, and a small bass amp.

We picked up grand pianos wherever we went—and, with temperature and humidity in Asia being what they are, it's hard to maintain a piano in good condition. Quite a few of the ones we used were not up to par. I found, after some experimentation, that the PZM worked best taped to the side of the piano near the top end strings. Kenny preferred to play with the piano lid open on the tall stick, and I could never get enough gain with the mic taped to the piano lid, where I normally use it. An AKG D-200E about 4-6-in. above the low and mid strings cross point proved sufficient for the low end. Some pianos sounded so "tubby" that it was necessary to dump low end both in the PZM and the D-200E. In these cases, or when I needed more gain, I would add another D-200E underneath the piano, positioned in the midrange area close to the hammers, no more than 6-in. from the soundboard. This could increase bass response without EQ, and contribute added tone and volume to the left hand for "comping."

EXPERIMENTAL MIC'ING

One of the most important aspects of Chico Freeman's playing is his warmth, both musical and personal. Our "triple-threat" mic'ing evolved out of experimentation to capture this quality. An EV DS-35 on a short boom functioned as a "bell" mic for tenor and bass clarinet; as a lower pad mic for soprano sax. Another DS-35 on a tall boom was positioned to pick up sound from the top pads of the soprano and bass clarinet. Upper notes on these instruments are much louder near the top than the bottom; double mic'ing achieves a smooth sound with no volume dropoffs due to, say, a quick leap from a low note to a high note and back again. A third DS-35 on a tall boom was used only for tenor, positioned towards the side of the tenor where the large pad is. Branford Marsalis was the first player I ever saw using this technique; it can give the tenor a wonderfully rich low end. The last DS-35, on a straight stand, served as Chico's announcing and flute mic. After a few gigs, Chico developed the ability to use the mic's proximity effect as a creative tool. Using his bass flute, Chico would suddenly decrease his distance from the mic, getting right on top of it while blowing certain notes. Due to proximity, the added bass response enveloped these notes, adding greatly to their power and richness. He would also place the flute mouthpiece directly on the mic and, without blowing, tap the pads to create bassy harmonic "pops" as a solo device, or to accompany others.

The rest of the mic'ing remained fairly standard. Wallace Roney played trumpet into a DS-35 on a straight stand. Clarence Seay used Underwood pickups on his string bass, and a Polytone Mini-Brute amp as a monitor. He preferred to be mic'ed, so I used a DS-35. If I needed an extra mic for a jam session, I used my D.I. box, post amplifier, using the padded input. Ronnie Burrage's Tama drum kit was mic'ed with DS-35's on the kick drum and snare/hi hat. AKG D-200E's were used for the two overheads.

The Gulf Hotel Ballroom was fairly small, seating about 250 for our concerts, and sounded fairly live despite the plush carpeting. There were two 15 amp, 220 volt, grounded receptacles of the new U.K. type in the sound booth backstage, with very steady voltage. The stage sound proved to be almost adequate to fill the room, with some piano and a pinch of sax and trumpet solos added to the PA. We were all a bit nervous, not only because it was the first show, but due to the musical climate in Quatar. The door had been closed on Western music, but we could play a key role in opening it again. The concert started slowly, but warmed up fast. Quataris may not have been exposed to much jazz, but after a few numbers I noticed quite a few heads bobbing and toes tapping. Our show the next night was even more successful—I noticed several repeaters from the previous evening, and the response was much more appreciative. It was nice to see that once familiar with the music, the rest came naturally.

The band and I also had an opportunity to visit the QBS (Quatar Broadcasting System) headquarters in Doha, to witness a recording session featuring some of the country's finest musicians. Once there, I was dragged through the facility by several enthusiastic Quatari engineers. The studios were spanking new, featuring the finest in modern European gear—Studer, Schlumberger, Neumann, and Sennheiser all being well represented. I was also invited into the control room by the producer to listen in on the mix. Monitor levels were LOUD all the time, and I noticed a 6 kHz boost on almost every input channel. The Egyptian/Quatari group consisted of several violins, wood flute, two hand drummers, zither, and a vocalist, who was the featured artist. The music was a wonderful collage of polyrhythms and sustained passages that contained many bent notes and rhythmically-shifted pitch. We, of course, invited them down to our concert, and several came. Afterwards, all were eager to share ideas and techniques in the international language of music. With a good feeling from sharing music, we prepared for our next destination—Pakistan.

NEXT STOP—PAKISTAN

The 19th found us flying from Doha via Dubai to Karachi, our first stop in Pakistan. Here, I felt like I was finally in Asia. Our concert was to be the next day at the Taj Mahal Hotel Ballroom. The facility was quite large—1400 seats, with a balcony, and contained a proscenium-type stage surrounded by a large apron that created a border of about 20-ft. between the stage and the first row of seats. There was also an in-house PA system, a flown center cluster that contained 4 JBL 15-in. loudspeakers in Perkins-type enclosures, and two JBL diffraction horns with JBL 2441 drivers. I decided to use it as a balcony fill and to augment my own PA. Daraz Kahn, the local USIA audio guy, was to provide a step-down transformer with stabilizers, so I wouldn't have to use mine. We planned on a 2:00 PM sound check, and I then enjoyed an evening of exploring Karachi.

Things on the 20th, however, did not go as planned. Ike called me in the morning to let me know that our gear still had not cleared customs. As it happened, Ike had to drive back out to the airport with Lee, and got our gear released by 3:30. Of course, the gear didn't arrive at the hall until 4:30. I hurriedly set up, and managed to get a 30-second sound check before doors opened at 6:00. I noticed that the in-house PA was incredibly bassy; my augment PA graphic channel showed 15 dB of cut at or below 500 Hz. By walking around the hall a bit I realized why: one of the horns was dead! No one at the facility had even discovered this; too bad I don't get paid for house calls. During the show, Chico got the audience to participate, having them yell at certain points of "MI-FI-TI," a composition by Ronnie Burrage.

The 21st saw us fly from Karachi to Islamabad, the capital city of Pakistan. We stayed in neighboring Rawalpindi, and prepared for our concert on the 22nd.

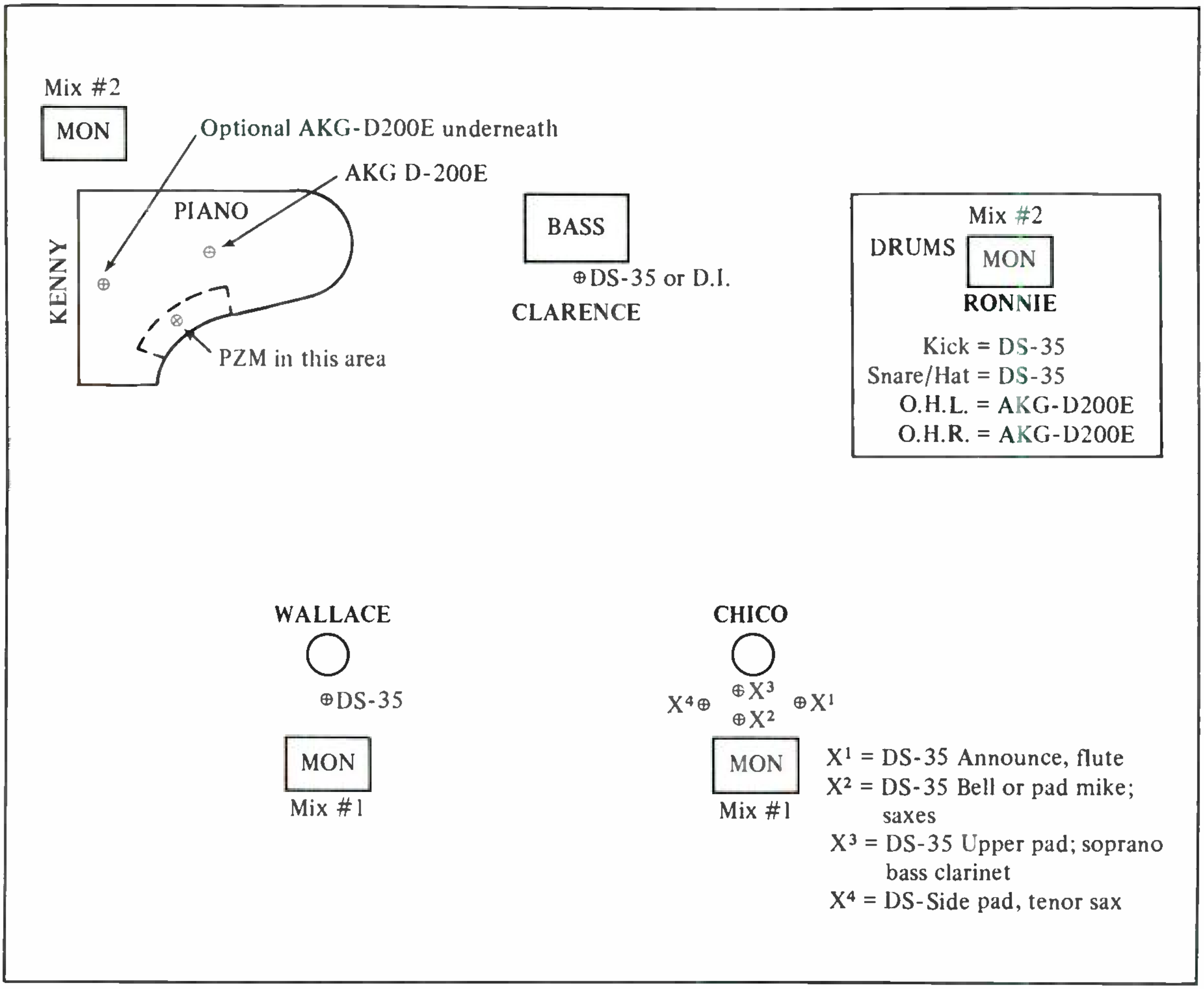


Figure 1. Stage set-up for the Chico Freeman Band.

This was to be attended by many important diplomats and Pakistani cultural officials, with the attendant protocol. Liaquat Hall seated about 900, on wooden tiers that rose very steeply away from the proscenium-type stage. This resulted in every seat sitting at a level above the stage, necessitating not only elevating but tilting the PA cabinets much more than usual. Pakistani radio was taping the performance for later broadcast, and requested an audio feed from me as well as setting up their own mics. There was plenty of power outlets stage right—five and fifteen amp, 220 volt, old U.K. receptacles. As it turned out, only one of these outlets had a functional ground. The back row of seats couldn't have been more than 75-ft. from the stage, so a restrained level was called for. The concert was preceded by a good 15 minutes of speeches from various Pakistani and U.S. officials, and the formal nature of the crowd resulted in a reserved audience. Chico wisely called sets comprised mainly of ballads and straight-ahead material.

Our next stop was Lahore, near the border of India. Everything was very lush and green and our concert on the 24th was held at the local arts center, across the street from our hotel. Its capacity was 750, in standard theater seating on a steep slope. Again, I found it necessary to tilt my top PA cabinet up for back row coverage. Thick carpeting, plush seats, acoustically-treated ceiling, and wood paneling combined to give this hall the best acoustics we encountered in Pakistan. There were plenty of fifteen amp, 220 volt grounded receptacles of the old U.K. type on the stage facing. I had a great time—everything sounded fabulous. The place was totally sold out—people were even sitting in the aisles and began their applause after the evening's first solo. Chico had decided to change the format of our shows in Lahore: Instead of two 45 minute sets with an intermission, one 90 minute set would be played. For this particular audience, the band played well over two hours, even playing a few requests from the audience. One other surprise: During the band introductions, Chico introduced me, and had me take a bow! I'm not used to publicity; an engineer is only noticed if the sound is terrible.

The 25th was a day off for the band, but a day of travel for me. Our last concert in Pakistan was to be in Peshawar, at the foot of the Khyber Pass near Afghanistan. The band and equipment were to fly from Lahore to Peshawar on the 26th, the day of our show. This aircraft was a Fokker Friendship, a prop-driven plane with a cargo capacity almost too small to handle our gear. I was to fly to Islamabad on the 25th with the gear, then drive the four hours from Islamabad to Peshawar in the

equipment truck the following day. Upon landing, the gear was loaded into the panel truck that would take us to Peshawar.

Our concert here was held at the Khyber Intercontinental Hotel Ballroom—a tiny room seating 200. I didn't even bother mic'ing the drums here; the venue called for the least amount of reinforcement on the whole tour. There were 15 amp, 220 volt, old UK receptacles, but none had a functional equipment ground. The house electrician and I discovered that the electrical conduit was grounded, so we ran a wire to it for ground. Our audience here was almost completely Western, mostly U.N. observers and experts working in the Afghan refugee camps. We were well received, but the size of the room, muggy weather, and bad ventilation made the room very saunalike—resulting in a short set by the group.

Before leaving Peshawar late on the 27th, the quintet did a performance for Pakistani television at the local station. The T.V. producer was concerned that his audio equipment might not be up to par, but it proved to be adequate—a 14-channel Schlumberger console, (EQ on the first nine channels only), and plenty of Sennheiser and AKG mics. I did end up using my PZM on the piano, causing quite a stir among the Pakistani engineers who had never seen one. In a three hour taping session we recorded an hour of material. Kenny took advantage of a good piano and turned in some spectacular playing. There were no monitors in the studio, so the guys had some trouble hearing themselves, but they managed to rise above it. As Chico remarked later, you never miss your monitor system until it isn't there. We then caught our afternoon flight back to Karachi, where we spent the evening before leaving Pakistan.

SOUND SYSTEMS IN INDIA

Our concert here was held at Rhang Bhavan, a 3000 seat open-air amphitheatre tucked away between streets. Niranjen Jhavari, our promoter, had contracted a PA system, as I had informed him that we would need a sound augment for a gig of this size. The venue is basically a fan-shaped concrete area, with the stage at the base of the fan. There is a series of concrete terraces to provide a slope for the seats. I agreed to supply monitors, a monitor amp, and mics. Everything else was supplied by the PA company. The system proved to be a mixed bag, some of which was apparently donated by the Police (the rock band) after they played Rhang Bhavan. It consisted of 15-in. Gauss speakers in folded horns, 15-in. Gauss in Altec Voice of the Theatre cabinets, Altec multicell radials with JBL 2441 drivers, and two boxes containing four JBL bullet tweeters each. Amps were four Crown DC-300A's. There was an electronic crossover with fixed points of 200 Hz, 1800 Hz, and 10 kHz, with level controls for each section. Initially the system sounded very harsh, but by turning down the tweeters and horns and turning up the mid bass, things smoothed out considerably. The concrete environment contributed to making things sound brighter than normal, as well as creating a nasty 1½-second slap echo on the stage. The band needed the highest monitor levels of the tour to compensate. I tried to minimize this echo by keeping the overall P.A. level down. This was a new concept for the Indian technicians, who ran at rock levels all the time.

This concert was special with, by far, the largest audience of the tour. Thirty-four hundred tickets were sold, and at least another 1000 crashed the gate, scaled the back wall, or sat on rooftops to watch the show. They were knowledgeable about jazz, and gave both rapt attention and thunderous applause. The band played an amazing 2½ hours to this adoring crowd, turning in some of the best music of the tour. The press loved it—the reviews dubbed it "the greatest jazz concert ever in Bombay." Quite a compliment, considering Duke Ellington, Sonny Rollins, and Stan Getz all played here. I was further amazed when the reviews also mentioned that, for the first time, music could be heard clearly and concisely at Rhang Bhavan. Both Mr. Jahvari and Roger Drego, the chief PA technician, took note of my different approach to volume, and mentioned that they would adopt this idea for future jazz concerts. Contributing to the improvement of future concerts meant the most to me.

CONDITIONS IN BANGLADESH

With the good feelings from Bombay, we left via Calcutta for Dacca, Bangladesh. We were to play two concerts, as well as perform a morning concert for Bangladeshi T.V. The ON/OFF switch of one of my Crowns had vibrated loose, due to a rough plane ride, so I had to short it ON permanently. Things were far from ideal at the venue either. The 1400 seat Shilpakala Academy Auditorium had a concrete slab floor, and a barn-shaped metal ceiling with open sides; a very hollow-sounding room, with a nasty 1 kHz resonance. I set up as far as I could, but had to wait for power. There were several 15 amp, 220 volt, old UK type receptacles, stage right, on a board, but none had an equipment ground. When I finally got a functional ground, from a water pipe stage right, I discovered that the neutral was carrying 50 volts! I pointed this out to the house electrician, who went to work on it. Meanwhile, Chico and the band had attended a special musical performance by musicians attending the academy; they were so impressed that they asked the young Bangladeshis and their instructor to join in the evening's performance. Most of that day's soundcheck now became a rehearsal, with everyone figuring out what to play. I put up all my available mics, and pulled some off Chico to mic the "expanded" quintet. There were three violins, two tabla drummers, and the professor on wood flute.

Naturally, in the middle of the first song all power totally died, plunging the hall into darkness and staying that way for about three minutes. Apparently these power outages are common in Bangladesh. The band played right through it, however, and had the crowd right with 'em. The voltage was all over the place: I observed swings between 100 and 135 volts on my power monitor, necessitating several trips to my transformer to adjust voltages. The addition of the local musicians was a huge success, as it showed Chico's respect for and interest in the local culture. And the audience showed how much they appreciated it by going crazy. The dean of the school remarked to me that he's never seen a crowd so enthusiastic. The T.V. producer stopped by for a chat after the show. Because their audio console was only 6 channels, I was to provide mixing, microphones, cables, and stands.

On the morning of the 7th, I returned to Shilpakala Academy to pick up what I needed for the taping, then went to the T.V. studio to set up. It went fairly smoothly, except it took 1½ hours to get a functional equipment ground. Grounded receptacles seem to be non-existent in Bangladesh, although I think I gained a few converts by getting rid of buzz in the audio. Once again, we also included the musicians from Shilpakala, now swollen

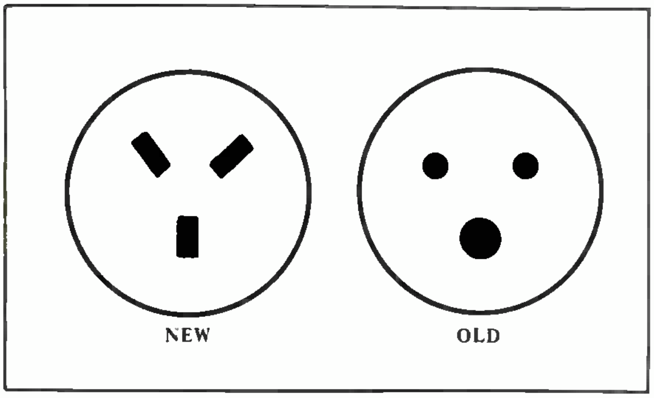


Figure 2. Power receptacles.

with eight violinists. Afterwards, it was back to Shilpakala to reset the sound system.

All seemed to be going well until about half an hour before the show, when a huge power surge occurred and fried my console power supply. The academy owned a Yamaha PM-1000 that operated on 220 volts, and they were kind enough to loan it to me in this time of need. Of course, I had to do the show cold, with a new console and no sound check. Somehow, I got through it, although the monitors were somewhat lost until the second song. The guys were real helpful, giving me hand signals to get our normal monitor mix happening. Once again, the combination of the quintet and Shilpakala musicians brought down the house. I was most concerned about getting my console repaired before leaving the next day.

I spent that next morning at the local American center trying to fix my damaged mixer. The problem turned out to be two blown fuses and a fried -15V DC regulator, all of which I had with me. My "afterthought" to include power supply parts had proved to be good intuition—not to mention saving my butt. Having solved that problem, I happily bade fairwell to Bangladesh, and was welcomed back to Calcutta, where performances were scheduled for March 10th and 11th.

POWER IN CALCUTTA

Two things make Calcutta a difficult place to work: the weather is extremely muggy, and there are often power blackouts. There isn't enough electrical power to meet demand in Calcutta, so a program called "load shedding" is employed; whole sections of the city are blacked out at designated times. Most businesses (and our hotel) had their own generators to supply power during the blackouts.

Our first concert here was held at the Dalhousie Institute, on a portable stage, under a large tent near the swimming pool. With folding chairs, the capacity figured to be around 1500. A 15 amp, 220 volt, old U.K. grounded receptacle on a board was run out to the stage from the Institute. When power worked, I observed fluctuations of 5-15 volts, though never any surges a la Bangladesh. A gasoline-powered generator was also provided to act as a backup in case of blackout. I found it necessary not only to provide a ground wire for this, but to ground the generator's neutral as well. We had another overflow crowd—USIA estimated at least 2000, not counting the people outside the tent, by the outside fence, and on top of the changing rooms, trying to see us. Again inspired, the quintet played for over two hours to this very responsive audience. The tent helped contain the sound somewhat, but for the crowd I had to deal with I was definitely underpowered—next time I'd ask for an augment here.

The second Calcutta performance was held at Gyan Manch, a small theater seating only 400. The small size, plush seats, carpeting, and acoustical treatment of the back wall combined to make the acoustics excellent: a manageable reverb time with even decay across all frequencies. Of course, the power was out again, but was restored at 2:00 PM, and stayed on. The local power commission agreed to maintain power to this section of the city throughout our performance as a goodwill gesture! Pockets in the stage right floor contained several 15 amp, 220 volt, grounded old U.K.-type receptacles. Once again I noticed those odd voltage drops. This concert had been oversold somehow; there were 200 people without seats, and even more crammed into the lobby trying to hear the band. Kenny agreed to play without a monitor speaker so we could attempt to accommodate these people. I took his cabinet and dropped it in the lobby, running it off my house PA amplifier. He didn't seem to miss it: the combination of good room acoustics and a good piano coaxed a stunning keyboard performance out of him. Chico used his flutes more at this show, with impressive results. Once again, the audience in Calcutta responded warmly to the quintet. Our next stop—Delhi, the capital of India.

OFF TO DELHI

Two concerts were scheduled in Delhi, both at Kamani Auditorium. It was a very modern auditorium with a capacity of 670. There was a balcony, necessitating a tilt in my upper PA cabinets for coverage. The side walls of Kamani were extremely reflective; I found that placing my P.A. further from the wall than normal, with a slightly inward tilt, helped decrease reverb time noticeably, especially in the low end. Power was stage right on an elevated platform, with 15 amp, 220 volt, grounded receptacles of both old and new UK-type receptacles, but I found some of these were wired out of phase.

Delhi proved to be a very relaxing spot for us. Both concerts went well. The first played to a primarily VIP diplomatic audience, the second to a younger crowd, with many local musicians in attendance. The guys also gave a workshop at the American Center that was very well attended. It was surprising to learn that the young Indian musicians who play Western-type music rarely play together outside of their own bands! The idea of a "jam session" was incredibly popular, and arrangements were made with USIA to provide a space where these young musicians could gather weekly and jam, trading licks and compositions.

UNEXPECTED ADULATION

March 17th found us on the road again bound for Colombo, Sri Lanka via Madras, India. We arrived in Colombo on the 18th, early enough to enjoy the tropical weather. During a reception for us that evening, we were treated to a performance by the famous drum troupe of Sri Lanka. Ronnie jumped up and joined in after a few numbers, speaking to his fellow drummers thru their own instruments. Together, they totally tore the place up. Naturally, plans were then made to include the drummers in our show the second night at the John de Silva Auditorium, where a large native crowd was expected.

Our first concert was held indoors on a portable stage,

located in the courtyard of the American center across the street from the mission of the U.S.S.R. I think I noticed a few people standing near the gate during our show, checking us out—musical detente, so to speak. The audience here was comprised of VIPs and Sri Lankans who were graduates or students of universities in the States. This proved to be the most reserved audience of the tour. Chico was mildly disappointed in the response, and played a shorter set as a result. When Wallace was featured on his trumpet solo, Chico even walked out front, to sit with me at the mix point to watch and listen. I had the normal anxiety attack, anticipating what he's say about my mix, but he made that very clear—he loved it! It's very gratifying to know that you're on the same wavelength as the artist, with respect to the sound of the group.

And I needed to be able to trust my own judgement, because the following day was a nightmare. John de Silva turned out to be a rectangular concrete and metal box with seats—1500 capacity. It had a horrendous 3½-second reverb time, especially bad in the low end: any kind of low frequency information rendered any other sounds unintelligible. It's very similar to Chicago's Rosemont Horizon. Fifteen amp, 220 volt, old UK-type receptacles are located on boards stage right, but none had functional grounds. I got my ground from a water pipe in the hall behind the stage. To try and deal with the sound, I had Clarence turn down the bass eq on his amp as low as he could, then added his instrument to the monitor mix, so everyone could then hear him. I rolled as much bass as I dared out of the P.A., and tried to keep bass and kick drum low in the mix. It was marginally successful, but I was never really happy with the sound. The drum extravaganza, however, was the hit of the evening. Ronnie and the drum troupe were featured on a percussion-only piece, and also jammed during the drum solo on Ronnie's composition "Mi-Fi-Ti." This brought the house down although towards the end the Sri Lankan drummers stopped playing, transfixed by what Ronnie was doing. He had ALL the pots on. Unlike the previous evening, this audience was very demonstrative. We received a standing ovation which, according to our hosts, is very unusual in Sri Lanka.

THE RETURN TRIP

I'm sure we all wanted to stay on this island paradise longer, but we had a tour to finish. Back to India, and Madras; the largest city in southern India, and a cultural center for not only Indian classical music, but the Indian movie industry as well. Madras was in the midst of a terrible drought. Water was in very short supply, and was occasionally turned off, as was the electrical power. Oh no, more load shedding. This delayed my set up on the 22nd, as I couldn't set up at the Music Academy until there were lights. This facility seats 1580, with half of them in a large balcony. Acoustically, it was very similar to Hill Auditorium in Ann Arbor, so I felt right at home. Both are very live, with plaster walls and ceiling. There was a nasty long reverb around 100 Hz, so keeping bass volume low was a must. By using the stage sound and keeping PA volume low, a good mix can be produced. Power here was available from a platform stage right—15 amp, 220 volt, old UK-type grounded receptacles. Once again we had an overflow crowd—200 people sat in the wings as guests of the band or USIA. I noticed that the voltage climbed steadily during the show. I had to turn down my transformer output about halfway through. Apparently, as demand for air conditioning goes down, the voltage goes up.

Our last concert was scheduled for the 25th in Bangalore, called the "Garden City" for its greenery. We arrived from Madras on the 24th, and proceeded to relax, not only to prepare for our last show, but also to plan for a workshop to be held in the morning, prior to the show. I normally didn't attend the workshops, as I felt they were primarily for musicians. Chico asked me to attend this one, as he had received requests to discuss sound reinforcement, and wanted me to speak on the subject.

As a result, I decided to set up the day before all this took place. The Chowdiah Memorial Hall was a very distinctive place, shaped like a giant violin. Chowdiah was a very famous Indian Karnatic violinist, hence the shape of the structure built in his honor. The theater inside was a standard rectangular shape with wood paneling, plush seats, and great acoustics. There was plenty of power stage left, with plenty of 15 amp, 220 volt grounded receptacles of both old and new UK types.

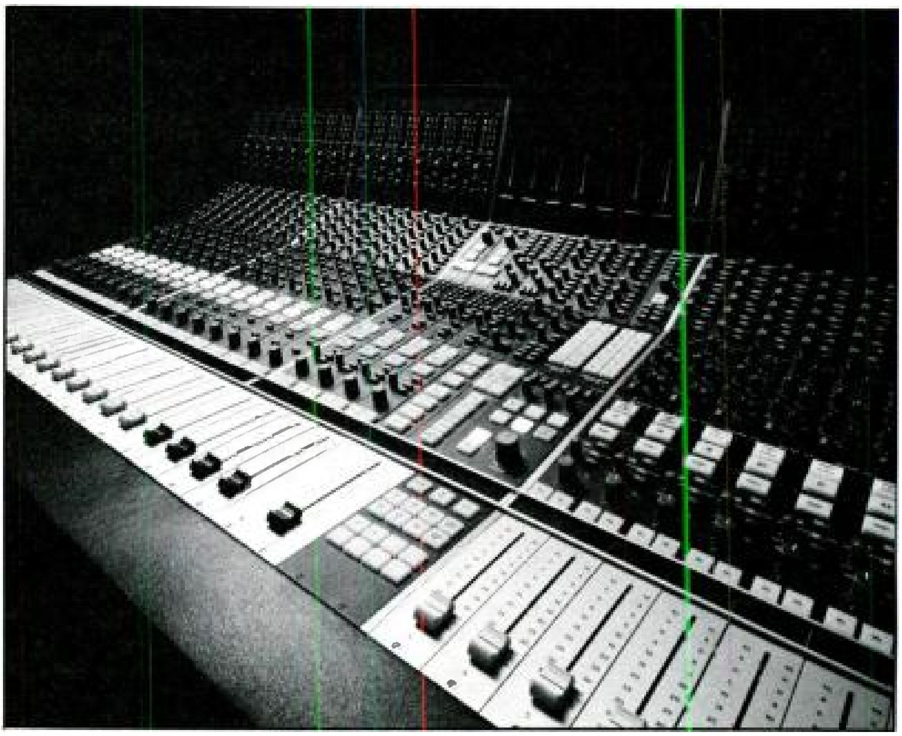
The workshop was very interesting, with both Chico and Kenny taking lead roles. Chico lectured for an hour on jazz composition and performance: Kenny gave a demonstrated history of jazz piano styles and influences. I went last. Only one person raised a hand initially, asking about Chowdiah's acoustics, but once I provided the answer, I was literally deluged with questions from the rest of the crowd. I attracted a large following once we split up into individual interest groups. I was asked the gamut of questions: from how to mic instruments to how my PA system was powered. Many club musicians wanted to know how to get their monitors louder without feedback, or how to use delay devices effectively. Overall, I was impressed by the awareness of sound reinforcement shown by these kids—although I didn't feel as though I'd even begun to answer all their questions.

I saw many of the people that had attended the workshop at the concert that evening. I found myself answering questions again, right up to showtime. Everyone sounded great, especially Kenny and Wallace, who burned up their featured numbers. Chico spoke to the audience for 20 minutes before the last number, publically thanking each member of the quintet, Ike, and myself. He also summed up how special it was to be able to tour India, and be touched by their music and hospitality. Thunderous applause told us how they felt about us and our music.

The 26th found us heading back to Madras, preparing to go our separate ways. Chico, Wallace, Kenny, and Ronnie were flying to Tokyo to meet bassist Cecil McBee, and do another 2 weeks of concerts in Japan. Clarence, Ike, and I were to return to the U.S. via Delhi and Frankfurt, Germany, a trip that would take us 2 days. After a series of smooth flights, we returned to JFK in New York on March 29.

I've only touched on some of the tour. Some other things I'll never forget—body surfing and laughing with Ronnie; Wallace, the international hamburger critic; playing basketball with Chico and Clarence; enjoying Indian food and talking music with Kenny; Ike cracking sick jokes and keeping it all together; working together and making six new friends. All the people around the world that were so nice to us, and so enthusiastic about the music. And, the compliments and good feelings from the quintet, the ultimate critics of my work. Best of all, I was given the ultimate compliment—more work. Once home, I began preparing for my next trip in the fall of 1983. ■

SOLID STATE LOGIC SL 5000 M CONSOLE



• Solid State Logic's new SL 5000 M series audio production system has been made by a new level of design at the basic component level. SSL has developed a new series of hybrid chips which replace the earlier op-amp-plus components type of sub-assembly. This array of hybrids substantially reduce the size, weight, complexity and power consumption of the system's building blocks. For example, a single SSL Hybrid Chip, providing a totally balanced and earth-free output capable of driving a 600 Ohm load at +28 dBm, takes up an area of about 40 mm by 15 mm by 5 mm. This is approximately one-tenth of the area previously required. To complement this hybrid technology, Solid State Logic has developed an entirely new generation of audio control desk architecture, which incorporates new materials, electronics, mechanical assemblies, and production techniques. All SL 5000 M Series consoles are based on a highly organized yet extremely flexible arrangement of audio, logic and data buses. A system of modular motherboards is used to distribute these buses throughout the console mainframe. A unique and important aspect of this arrangement is that the audio and control buses are distributed both horizontally and vertically. The 36 standard mainframes can be visualized as having a number of positions, which are the vertical channels running from the fader through to the penthouse. Each position is 40 mm wide, and is divided into a number of rows, which are the horizontal slots running across the mainframe. Each row is 150 mm high, with the exception of the penthouse row, which is slightly higher. SSL has developed an initial family of 28 Eurocard audio and control cassettes for the new series. The miniaturization afforded by SSL Hybrid Technology has made it possible to include complete control and data circuitry along with specialized audio electronics in each of the 40 mm by 150 mm cassettes.

A unique and important aspect of this flexibility is that all of the switching within the console is totally electronic, and that the vertical and horizontal bus arrangement provides address lines to all cassettes, regardless of their individual positions. The SL 5000 M Series may be fitted with mono and stereo inputs. The range of cassettes include mono and stereo faders, aux sends, equalizers, compressor/limiters, expander/gates. Depending on the mainframe size, each mono and stereo channel may be fitted with as many or as few of these facilities as is desired. The SL 5000 M Series will be available with two levels of computer assistance. The first level is known as SSL Instant Reset. This computer will store all switch settings, allowing broadcasters to instantly reset the console between any number of master and local configurations. All SL 5000 M Series Channel Cassettes are also addressable by the SSL Total Recall system, allowing the exact values of all variable controls to be stored and recalled with a control accuracy of 0.25 dB. The SSL Studio Computer also interfaces with the SL 5000 M Series, providing complete dynamic mixing automation and machine control. Thus, the SL 5000 M Series will be data-compatible to a large extent with all SL 4000 E Series Master Studio Systems and SL 6000 E Series Stereo Video Systems. The SL 5000 M Series has been specifically developed to provide the broadcast community with a means of standardizing on the most advanced audio technology at all levels of operation. Each department may custom specify their console to suit the precise requirements of individual control rooms. Yet throughout the plant, all consoles will share the same operation characteristics and maintenance routines. This dramatically reduces training and orientation time for the mixing and technical staffs. Spare parts stocks are common to all consoles, simplifying administration as well. Of course, this same versatility means that SSL can purpose-build an SL 5000 M Series system to suit the special needs of the independent production house or the local broadcaster. The complete range of SL 5000 M Series Audio Production Systems will be detailed at the National Association of Broadcasters convention in March of 1985.

Mfr: Solid Stage Logic

Circle 31 on Reader Service Card

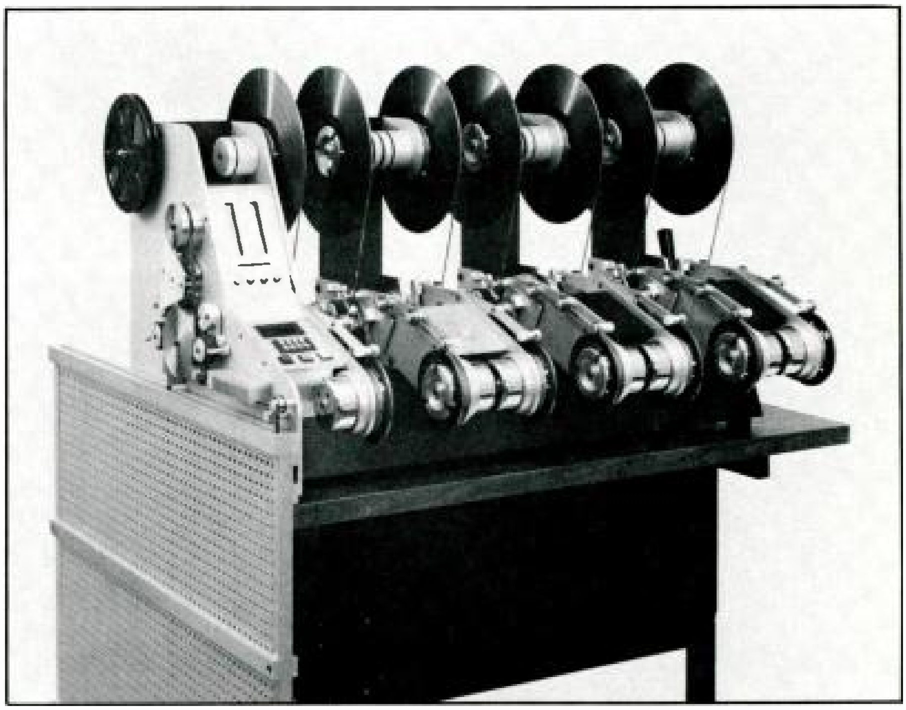
EVENTIDE H969 HARMONIZER

• Eventide's new harmonizer special effects unit, the H969, employs a newly designed digital intelligent splicing algorithm system (dubbed ProPitch by the company) which delivers pitch change performance without glitching over a wide frequency range. Eventide has also used a 16 BIT PCM linear coding for the first time in a harmonizer unit, in the H969. This unit has several new features that will be of particular interest to musicians. A dozen pitch-change presets have been included, enabling the user to instantly set a precise minor third, major third, fifth, seventh, or octave of pitch change. Each can be selected as a sharp or flat. In addition, separate coarse and fine adjust controls enable the user to set precise and stable pitch ratios easily. Full bandwidth delay has been increased in the H969 to 1.5 seconds, with a further increase to 3 seconds available at half bandwidth. The user can choose and save any five delay times for instant recall. The full delay range is also available in repeat and reverse modes. Delay time and pitch ratio are displayed on independent readouts. Doppler effects have been added to the H969 and flanging is also available. The H969 is said to be an addition to the Eventide family, not a replacement. The H949 will still remain popular, and Eventide says that the H949 is still better suited for time compression, broadcast and commercial production.

Mfr: Eventide, Inc.

Circle 32 on Reader Service Card

MAGNEFAX TAPE DUPLICATOR



• The Magnefax model 7574 represents the latest addition to the Magnefax line of high speed tape duplicators. Featuring the common mandrel design which has been used on all Magnefax duplicators since 1959, the 7574 also incorporates some of the most needed features for high quality tape duplication. Digital peak meters with memory hold provide pertinent level information throughout the entire bandwidth even during the shortest musical passages. The total low level audio path from the playback heads has been shortened to the minimum while the use of selected components and plug-in amplifiers insures total immunity from noise and high-end dropouts. The tone injection module offers a wide range of level and frequency adjustments to provide total flexibility when dealing with the loader requirements. A digital counter on the control module provides the user with the number of tapes recorded per slave and the machine will stop automatically when the preset number of passes has been reached. (The STOP function is actually performed in two steps in order to leave enough tape at the end of the pancake for easy threading of a loader without losing the first cassette.) All of the commutation circuits make use of optocoupling technology for minimum noise and increased reliability, and power switching is accomplished with the help of zero-crossing devices to reduce turn on/off EMI to zero. The loop bin will accommodate up to 1800 feet of ¼ in. master recorded at 7.5 ips. The conductive nature of the material used for the unit contributes to the absence of static electricity (the open design also eliminates the problem associated with air compression which often result in poor tape handling). The three point plates used for the record heads provide perfect support and stability and minimize the interactions between the adjustments, the heads are long life/high output Permalloy for best headroom and low noise. Dual synchronized bias oscillators are used for optimum crosstalk rejection and frequency stability. The bias output is shown on a bargraph for easy reading and a small loudspeaker is provided for monitoring purposes. The 7574 will produce in excess of 6900 C-45's/24 hours day using a 16:1 duplicating ratio for optimum quality and low maintenance. Frequency response is 30 Hz to 15 kHz ±2 dB, crosstalk from A to B is more than 55 dB and signal to noise ratio is within 2 dB of bulk erased tape.

Mfr: Magnefax International, Inc.

Circle 33 on Reader Service Card

People, Places

• **Dennis Ray Kirchhoefer** has recently joined the **Electro-Voice** engineering team as engineering project manager/microphones. In his new position, Kirchhoefer is responsible for defining and developing new microphone products and product line concepts, according to Watson. Prior to his appointment at EV, Kirchhoefer was a project engineer at Shure Brothers, Inc. While there, he developed or assisted in the development of a variety of general purpose and professional microphones.

• **Keith Odle** has been named as second engineer. Prior to joining the Castle staff, Odle was an assistant engineer at Emerald Sound Studio and Bullet Recording Studio in Nashville. He has also worked as a technical maintenance engineer at Soundstage in Nashville and the Record Plant in New York. **Gil Reaves** has joined the staff as synthesizer programmer/engineer. His previous audio experience was as an assistant engineer to Marshall Morgan. Newest addition to the Castle staff is **Tommy Dorsey** as synthesizer programmer and arranger. Dorsy was a piano performance major at Belmont College in Nashville and has worked as a keyboard player and session musician in Nashville studios. The Castle Recording Studio is located on Old Hillsboro Road, Route 11, Franklin, Tennessee.

• **Robert Dupras** has been appointed acting regional distribution manager for the Burbank, California, Regional Distribution Center of **Agfa-Gevaert, Inc**. Mr. Dupras has been with Agfa-Gevaert since 1966 and has worked with them in the Physical Flow areas and most recently served as office manager of the Burbank facility. His previous experience has been as assistant operations manager for Royce Photographics, an Agfa-Gevaert dealer in Glendale, California.

• **Garry Templin** has been named central regional sales manager for **Electro-Voice, Inc.**, a supplier of high-technology microphones, speakers and electronics to professional and consumer markets worldwide. Most recently, Templin was vice president of Cambridge Marketing Group, an electronics rep firm in Westerville, Ohio. Prior to that he was associated with EV as a sales representative for C. L. Pugh & Associates, and had earlier retail experience as sales manager for Audio Warehouse of Pennsylvania. EV is a subsidiary of Gulton Industries, Inc., a New York Stock Exchange company with corporate headquarters in Princeton, New Jersey.

• **Datatek Corp.** has appointed **Daniel F. Antonellis** as director of marketing and sales. Mr. Antonellis' function will include all marketing and sales activities for the corporation. Datatek is leading manufacturer of video-audio routing switchers.

• **Richard M. Wolfe** has been named vice president of technology for Satcorp, Inc. In his new capacity, he will review innovations and trends in audio, video and satellite technology, with an emphasis on high definition video applications, to determine their potential for investment or developnent. Wolfe is the former president of Wolfe radio and television stations in Columbus, Ohio and Indianapolis, Indiana. He has also served as vice president of operations and engineering for the PREMIERE pay television network, based in Los Angeles, and as vice president of engineering and video technology at Twentieth Century Fox, also in Los Angeles. He is currently a member of the Society of Motion Picture and Television Engineers "Working Group on High Definition Electronic Production," for which he chairs the sub-group on Aspect Radios. Satcorp is a privately-held investment firm based in New York, with interests in satellite communications and video technology.

• **James F. Woodworth** has been appointed national sales manager of **CompuSonics Corporation**, marketer of the world's first series of professional and consumer, floppy disk based, digital audio recording systems. As the company's first national sales manager, Woodworth will develop and implement a comprehensive national and regional sales program for the CompuSonics line of professional and consumer products. Woodworth brings over 20 years of experience to his new position at CompuSonics. Most recently, he served as vice president and national sales manager at Audiotronics Inc., in Memphis, Tennessee. Before this, he was national sales manager at Studer ReVox America Inc., based in Nashville, Tennessee.

• **Bose Professional Sound Systems** has appointed **Vector Corporation** as representative for its line of sound reinforcement equipment in the Pacific Northwest. Vector, with more than 20 years experience in the professional sound industry, will be responsible for Oregon, Washington, and parts of Montana and Idaho. Lew Garling is president of the firm, located at 2401 Tenth Avenue in Seattle, Washington. Sales Representatives Hal Kephart and Dean Standing will handle the Bose account. Bose Corporation's Professional Sound Systems division manufactures loudspeakers for use in a wide range of installed and portable sound reinforcement applications.

• **Dolby Laboratories Inc.**, announced today a new title and increased responsibilities for **David Gray**. His new title is chief engineer of the Hollywood motion picture division. In addition to Gray's current responsibilities which cover technical liaison with film studios, supervision of Dolby Stereo licensing activities, his new responsiblities also include overall management of technical operations of the Hollywood office. He will continue to be located at Dolby Laboratories' offices in Los Angeles.

Television Conference

• The Conference on Stereo Television, the first industry meeting devoted exclusively to the topic of stereo and second-language TV sound, will be held at the Hyatt Islandia Hotel in San Diego, March 11th and 12th, 1985. For two days, conferees will study in detail the opportunities offered by these new video enhancements, as well as the many programming, marketing, production, and engineering issues raised by multichannel sound. Specific workshops during the conference will focus on how audio and video are becoming integrated in the viewer's home, on sources of stereo programming and the plans of the networks, on the problems encountered in converting a television station's plant to stereo, on creating effective joint promotions with set manufacturers and dealers, on the role of cable in delivering high quality sound, and on the audience impact of reaching the Hispanic market with bilingual programs. The Conference on Stereo Television is sponsored by Waters Information Services, Inc., and will be moderated by the firm's president and editor in chief, Dennis P. Waters. Waters Information Services, Inc., based in Binghamton, NY, publishes *The Stereo TV Report*, *Satellite Audio Report*, *NEW RADIO Cable Audio & Pay Radio Report*, and other industry newsletters.

Lowell Sound Recording Minor

• The University of Lowell's College of Music announced the addition of a minor in sound recording technology for Electrical Engineering Majors, to complement the complete revision and upgrading of its recording degree, the Bachelor of Music: Emphasis in Sound Recording Technology. The revised curriculum for primary study in SRT includes intensive work in musicianship, sound recording technology with hands-on production utilizing state-of-the-art equipment, video production, electrical engineering, acoustics, physics, mathematics, and directed general education. The BM: SRT degree prepares students for entry-level positions in all disciplines of the recording industry requiring musical abilities and critical listening. The minor in SRT for Electrical Engineering majors prepares students for the non-musical careers of the industry. Emphasis is on repair and maintenance techniques, equipment functions and operation, aural perception of equipment functions, and research and development. The minor complements the strong education afforded by the University's Department of Electrical Engineering, amalgomating practical applications with a solid theoretical understanding of advanced EE principles. The College of Music is also furthering its commitment to education for the recording industry through the exploration of graduate programs and continuing education for professionals. For additional information contact: College of Music, University of Lowell, Lowell, MA 01854.

New NY Music Center

• Kaufman Astoria Studios is being positioned as the new Music Center of New York, spurred by the February 1985 opening of Master Sound Astoria, a state-of-the-art, 48-track recording studio as announced by Ben Rizzi and Maxine Chrein. Rizzi and Chrein, joined by studio head George S. Kaufman, announced the venture and the reasoning behind a commitment of $3 million to its realization at a press conference held in the KAS Commissary. "New York is synonymous with the arts and Kaufman Astoria Studios and Master Sound Astoria are destined to play major roles in expanding the links between music, film, and video," they noted. Master Sound Astoria, located in the central building of the 13-acre complex, will occupy over 14,000 square feet of recording space. It will be one of the largest and most advanced recording facilities in the country. MSA will open with two recording stages, one geared for film scoring and large enough to accommodate a symphony orchestra, and one to handle virtually every other type of session. Additional recording stages and a mixing theater are planned. Adding further to the capability of Master Sound Astoria as a music center is the pre-wiring of three KAS sound stages—Stage E, 26,040 square feet; Stage G, 12,000 square feet; and Stage H, 12,000 square feet. Each stage will have 100 microphone inputs with monitor communications, both audio and video. The studios will be among the few in the world to feature "ABSOLUTE POLARITY." Studio A1 will be equipped with a full pipe grid system for video lighting enabling live shoots. An audio post-production area for video and film is also being installed. The SMPTE-equipped Control Rooms will provide computer mixing video post-production, a full compliment of outboard signal processing equipment, and virtually all track formats...both analog and digital. Rizzi/Chrein come to Master Sound Astoria with impressive credentials. Ben Rizzi has been a professional musician (keyboards); while Maxine Chrein has been a singer and is experienced in business affairs, financial planning and marketing.

Classified

FOR SALE

EQUIPMENT FOR SALE: NEOTEK Series III; 28x24 with 16 Busses, 6 echo returns custom "pre-return" submaster fader, extra patch bay, producer's desk, leg set, patch cords, spare parts, excellent condition; WHITE 32-band EQ's $550 ea. (312) 864-4460.

SENSURROUND LOW FREQUENCY ENCLOSURES with 18" CER-VEG 300 watt die-cast 96 oz. magnet driver. $195.00. **Empirical Sound, 1234 East 26th Street, Cleveland, OH 44114. (216) 241-0668.**

Used recording equipment for sale by major midwest studio. CALL Jim M. for prices at (614) 663-4030. **Consoles:** Sound Workshop Series 30 24x24, Sphere Eclipse 20x16, Allen and Heath 16x8, Logex-8 12x8, Interface 104 16x8 stage monitor mixer. **Tape recorders:** MCI JH-110 16 track, Otari MK III ½" 8 track, 3M M79 ½ track, Otari 5050B ½ tracks, Tascam 122 cassettes. **Other goodies:** Yamaha digital reverb, amp racks with cases, Great British Spring reverb, Ashly SC-70 crossover, Tapco 4400 reverbs, Bose 802 PA speakers.

FOR SALE: **Complete 24 track pro studio.** Turn key. Includes Grand Piano. $75,000. Free installation & training. **Call (503) 726-1378** after 6 pm or **(503) 998-2758.**

FAIRCHILD Portable console, 16 in/8 out with stereo monitor bus. Works off internal nicads or ext. A.C. power supply. $1495.00 O.B.O. **Call (201) 446-2976.**

Closing date is the fifteenth of the second month preceding the date of issue.

Rates are $1.00 per word with $25.00 minimum. Boxed ads are $40.00 per column inch. db Box Numbers are $8.50 for wording "Dept. xx" plus $1.50 for postage and handling.

Discounts: 6x—15%, 12x—30%.

ALL CLASSIFIEDS MUST BE PREPAID.

Send copies to:
db The Sound Engineering Magazine
1120 Old Country Rd.
Plainview, NY 11803
att: Classified Dept.

1984 INDEX

INDEX BY AUTHOR

INDEX BY TITLE

INDEX BY COLUMNIST

BARRY BLESSER

JOHN BORWICK

JOHN EARGLE

JESSE KLAPHOLZ

KEN POHLMANN

INDEX BY SUBJECT

ACOUSTICS

AUDIO/VISUAL

Improved Sound for Filmmakers Using One Pass System. Steve Michelson. Oct., 1984, p. 24.

Soundmixer's Guide to Videodisc. Charles P. Repka. June, 1984, p. 28.

VCA Teletronics Makes a Special Delivery. Mark Trost. Jan./Feb., 1984, p. 46.

BROADCAST

Digital Audio Broadcasting at WGBH, Boston. Peter Swawnson. Apr., 1984, p. 34.

VCA Teletronics Makes a Special Delivery. Mark Trost. Jan./Feb., 1984, p. 46.

CONVENTION REPORTS

NAB 1984—An Impression. Burrell H. Hadden. June, 1984, p. 38.

Tenth International Broadcasting Convention. John Borwick. Oct., 1984, p. 34.

CONSOLES

Recording Studio Consoles and Film Production Part II. Gregory Hanks. Jan./Feb., 1984, p. 34.

DIGITAL AUDIO

Digital Audio Broadcast at WGBH Boston. Peter Swanson. Apr., 1984, p. 34.

HISTORICAL

Brief History of the Loudspeaker. Jesse Klapholz. Aug./Sep., 1984, p. 32.

Father of Sound Systems. William C. Matthews. May, 1984, p. 43.

Golden Age of Radio. Skip Pizzi. June, 1984, p. 42.

INTERNATIONAL AUDIO

Audio Visual Center Wisseloord. B. Baars. May, 1984, p. 28.

MICROPHONES

C-duced. Jesse Klapholz. May, 1984, p. 35.

Development of a New Artificial Head. Stephen Peus. June, 1984, p. 34.

New Concepts in Microphones, The Age of the Miniature Cardioid Electret Condenser. Robert F. Herrold. July, 1984, p. 24.

RECORDING TECHNOLOGY

Ambisonic Mixing—An Intro. Richard Elen. July, 1984, p. 28.

How to Reduce the Risk of Producing On-Location Events. Joe Coencos. Mar., 1984, p. 30.

SOUND REINFORCEMENT

Band-Aid, Repairs on the Road. Bob Anthony. Mar., 1984, p. 26.

How to Reduce the Risks of Producing On-Location Events. Joe Coencos. Mar., 1984, p. 30.

Modular Sound Meets Its Waterloo. George Stephen. Mar., 1984, p. 40.

On the Boardwalk. Jesse Klapholz. Mar., 1984, p. 44.

Sound on Broadway. Susan Borey. Aug./Sep., 1984, p. 46.

Sound on Broadway Part II. Susan Borey. Oct., 1984, p. 40.

STUDIOS

Dallas Sound Labs. Martha Fischer. Mar., 1984, p. 20.

Recording Studios of Buenos Aires. Alberto Escrina. July, 1984, p. 36.

TAJ Soundworks Rock-n-Roll to Foley...And the Cable Link. Jim McCullaugh. Oct., 1984, p. 28.

TEST REPORTS

Problems With CD's—Real or Imagined? Len Feldman. June, 1984, p. 24.

TEST REPORT: Shure FP31 ENG Mixer. Len Feldman. Apr., 1984, p. 54.

TEST REPORT: Ursa Major 8X32 Programmable Digital Reverb Unit. Marco Fratnik/Leslie Shapiro. Jan./Feb., 1984, p. 54.

TESTS AND MEASUREMENTS

From 15 Millimaxwell to 1200 Nanowebers. Josef Dorner. July, 1984, p. 34.

Monitor Loudspeakers. John Eargle. Aug./Sep., 1984, p. 36.

Seeing What You Hear. Jesse Klapholz. Jan./Feb., 1984, p. 42.

Testing...One, Two, Three. Jesse Klapholz. Apr., 1984, p. 37.

Using the Tecron TEF System 10 Acoustic Analyzer. Bruce Bartlett. May, 1984, p. 32.